Technical Specification Group Services and System Aspects
All-IP Workshop, Nice, France, 7-9 February 2000

AIP(00)0021

Draft 02
Minutes of the workshop/tutorial between CN R00 ad-hoc, S2 and T2 on Release 2000

31st of January 2000
Puerto Vallarta, Mexico

A GLOBAL INITIATIVE

Chairman: Teuvo Järvelä, Nokia
Meeting Secretary: Alain Sultan, ETSI

Meeting Report draft 02

Workshop/tutorial between CN R00 ad-hoc, S2 and T2 on Release 2000

31 January 2000

1 Opening of the meeting

This one day meeting took place on Monday 31st of January. It started at 9 a.m. and ended at 5 p.m. It was chaired by Mr Teuvo Järvelä from Nokia, chairman of TSG SA WG2. The meeting was hosted by ATT, BellSouth, Ericsson and Nokia in Puerto Vallarta, Mexico. Mr Steven Hayes, from Ericsson, vice-chairman of TSG CN, welcomed the participants and gave some practical information.
The support for the meeting, including the redaction of these minutes, was provided by Mr Alain Sultan, MCC.

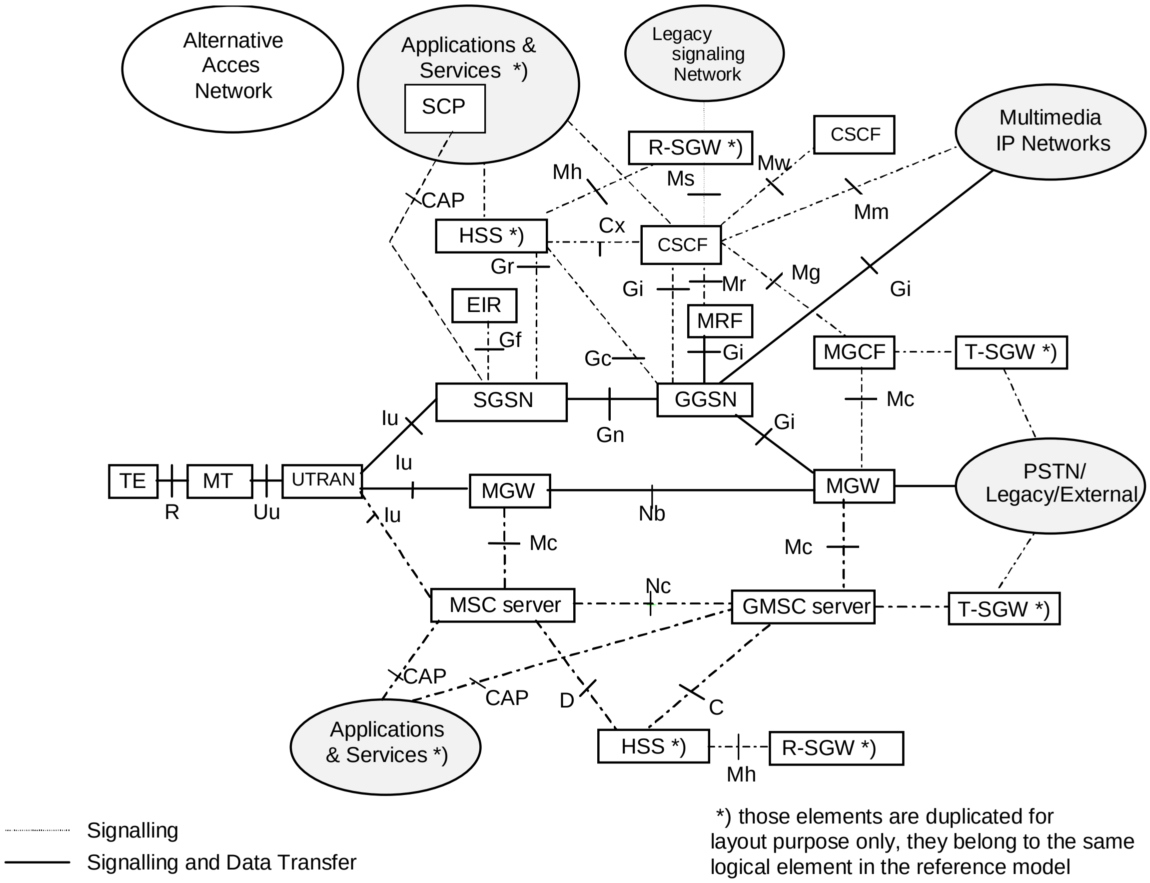
2 Presentation of the tdocs

S2K00-001, source SA2 chairman: *draft Agenda*
The agenda is provided as a part of the CN R00 ad-hoc meeting agenda, under the section "Joined workshop/tutorial with S2 and T2". It was approved as such.
Conclusion: Noted.

S2K00-003, source SA2 chairman: *general presentation of SA2 work on R00*
This document is a presentation from SA2 chairman on the work performed by this group on Release 2000 Work in general.
He remembered the principle and structure of the Inter-Group Co-ordination ad-hoc groups (6 IGCs, each one on a dedicated task, e.g. Services and Service platforms), which will be still used for R00, potentially modified.
He stressed that the actual work from S2 on R00 is presently contained in TR 23.821: this TR collects the SA2 R00 architecture related decisions. In a second step (starting between March and May 2000), this TR is intended to be closed without publication, and CRs will be provided against the TSs.
No discussion. Noted.

S2K00-002, source Ericsson and Nokia: *Release 2000 Architecture*
This is a presentation from Ericsson and Nokia on the status of the work of S2 from a technical point of view. It highlights the main conclusions contained in TR 23.821. The chairman explained that the source is Ericsson and Nokia not S2 because, due to lack of time, S2 was unable to review this presentation (main of the S2 decisions were taken on the week before, so the document was elaborated during the week-end).
Note: During the meeting, the discussions on this tdoc took place section by section. Such structure is repeated in the following text: "***Presentation:***" refers to what was said by the orator and "***Discussion:***" to the comments and questions made by the audience on each section.
Presentation: The paper explains that two major steps have been identified for R00: call and bearer separation for the CS domain and addition of "IP based Multimedia Domain" (IM domain) as an overlay to the PS domain.
Discussion: The foreseen impacts on the air interface on R00 are mainly the following: the terminal shall be able to support the potentially new call control, the real time performance of PS domain might need to be enhanced, and some changes might be needed in the UTRAN to support the traffic more efficiently (e.g. header compression). But the lower layers of the radio interface should not be deeply impacted.
It was noticed that there is an extension of the concept of "domain": primarily, "domain" was just used to distinguish between the CS and the PS domains. A "domain" is now any grouping of network elements which are often refereed to together (it is simply used to ease the discussion). It is explained that the R99 domains (CS and PS) are kept, there is just an addition of the "IP based Multimedia Domain".

Presentation: A general description of all the elements involved in the R00 architecture is provided. This architecture is the following:



The "key elements" are:

- the Call State Control Function (CSCF), which is the CC entity,
- the Home Subscriber Server (HSS), which is the master database for a given user (it contains the HLR functions plus some new functionality),
- the Media Gateway Control Function (MGCF), which mainly performs protocol conversion between the legacy (e.g. ISUP, R1/R2 etc.) and the IM domain call control protocols.
- the Transport and Roaming Signalling Gateway Function (T-SGW, R-SGW), which perform signalling conversion at transport level
- the Multimedia Resource Function (MRF), which performs multiparty call and multi media conferencing functions.
- the Media Gateway Function (MGW), which is PSTN/PLMN transport termination point for a defined network and interfaces UTRAN with the core network over Iu.
- the MSC server, which mainly comprises the call control and mobility control parts of a GSM/UMTS MSC.
- the MSC Server, 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. It also contains the VLR functions.
- and finally the Gateway MSC Server, which mainly comprises the call control and mobility control parts of a GSM/UMTS GMSC.

Discussion: The CN vice-chairman mentioned that a layered structure, and/or a model reflecting more clearly the domain split will be of great help for his group. So the domains should appear clearly in the figure, as to clarify the relationship between the domains and the entities. There is some ongoing work at SA2 on this issue.

In R99, there is a separate registration for CS and PS domain. It was explained that for R00, there is a working assumption at S2 that there will also be a registration to the IM domain.
As the HSS is a superset of the HLR, it provides at least all the functions the HLR provides, i.e. it maintains the location information, handles the subscriber information data and run the service logic in conjunction with the GMSC for the handling of MT calls.
It was explained that the separation of call and bearer applies both to CS and PS domains. Some doubts were expressed by NTT on the ability of the architecture to provide a single common "transportation level" using indifferently CS and PS domain. It was answered that the GGSN and the MGW might be implemented together (and the SGSN and MGW also) to stress there is a common "bearer provisioning platform", but this is not a requirement.
It was stressed that the reference model describes the logical functions, and then it can be chosen at implementation to group them in a single physical entity or to have all of them in separate physical entities.
The subject of multiple providers providing specific services to a given user has not been handled by S2. It might be so that the HSS shall contain the information related to all the providers linked to the user, but, again, this is not fully defined.
Presentation: The reference points between these entities were also presented.
Discussion: Concerning the Gm Reference Point (CSCF – UE), it is explained that UE is the combination of TE and MT (UE is presently not shown in the model). The Gm is not shown is the reference model because forgotten, and this will be fixed in the next version of the TR. However, NTT stressed that this one is different in nature with the other ones: all the other ones seem to be actually interfaces between contiguous physical entities, and this is not the case for Gm, which refers to a protocol running between CSCF and UE used on top of a set of interfaces.
This leads to some considerations on the difference between interfaces and reference points, with no real conclusion.
S2 has not yet allocated the task of the interfaces definition to the other groups. This will be done starting in March, with the help of the IGCs.
It is explained the Mg (between CSCF and MGCF) is mainly used for CC purposes.
The principle of having MAP/CAP being able to be run on different sets of transport layers is presented (MAP/CAP+TCAP+SCCP can be e.g. run on top of STM, IP, ATM,...).
Concerning the interface between the HSS and "Application and Services", not named on the figure, it is explained that for R99, between HLR and Camel server, the MAP protocol is used.
The level of stability of the interfaces was clarified: the bottom part of the reference model (MGW, etc., up to SGSN, GGSN) are quite stable. The part CSCF, MRF, MGCF, etc are less stable.
Presentation: Finally, the most urgent issues to be solved for the R00 architecture were stressed: the definitions of the domains need to be refined, the split of functionality between the visiting and home networks is not clear, what are the identities of the subscriber and the aliases of the terminal (Addressing Principles), and what are the CC and/or multimedia protocol(s) to be used for R00.
Discussion: Concerning the statement " Where is the IP address allocated? ", T2 chairman stressed that the IP address is located in the GGSN in GPRS, so if it is changed for UMTS, it should be for a clearly-identified reason.
Concerning the CC protocol(s), e.g. H.323 or SIP can be used by 3GPP, even if not defined by 3GPP. They can be re-used as defined by IETF.

S2k00-006, source S2 and Nokia, *R99 UMTS Quality of Service (QoS) Concept and Architecture*
This presentation is identical to the one made to N2 (approved by SA2). Only the last page ("Key issues for R00") containing the QoS key issues for R00 identified and approved at the previous SA2 meeting has been added.
The presentation provides the QoS aspects for R99, as defined in TS 23.107. The work has not really started yet on R00 QoS aspects at S2, but will be based on the R99 aspects.
The key new features for R99 are:

- Multiple PDP contexts with the same PDP address are supported.
- The concept of "Traffic Flow Templates" has been introduced. It allows to multiplex packets at GGSN: the TFT is the information available for the GGSN for multiplexing of downlink data packets onto several secondary PDP contexts, i.e. information used to select the right PDP context for a data packet.

- The UE is now able to control PDP context activation/modification and TFTs.
- New QoS parameters have been defined and now the same set of parameters applies for PS and CS domains. Four "UMTS traffic classes " have been defined: "conversational class" (e.g. of application: voice), "streaming class" (e.g. streaming video), "interactive class" (e.g. web browsing) and "background" (e.g. background download of e-mails). Some QoS parameters are not applicable to a specific traffic class: this is specified in 23.107. The list of possible values for each parameter is also provided.

Moreover, 23.107 also defines the QoS management functions (e.g. policing, monitoring, packet classification, marking, etc.) as well as their location in the network (i.e. which network element performs which function). It finally provides the mapping of the UMTS Bearer Service parameters onto the different interfaces and layers.

Discussion: the relationship between slide 3 (mapping of functions to entities) and slide 7 (allocation of functions to entities) should be enforced.

It was commented that some further work should be performed on the mechanisms used by the UMTS network to adjust the value of the QoS parameters used within the network to the value of the end-to-end QoS parameters.

S2 agreed the week before to have some co-operation with Tiphon on QoS.

It was questioned if the traffic classes were simply implicitly defined by the value of the parameters, or if it was explicitly established. No clear answer was provided.

Presentation: The key issues on QoS for R00 were identified as being the definitions of the following aspects:

- End to end QoS negotiations and provision (e.g. delay)
- Applicability of QoS negotiation mechanisms (e.g. RSVP signalling)
- Interaction between CC/application signalling and QoS negotiation
- Adapt a QoS policy framework for UMTS
- QoS requirements to carry application and control signalling (e.g. CC/Multi-media signalling)
- Ongoing support for inter SGSN change (handover and SRNS relocation), both within GSM and 3G and between GSM and 3G, to satisfy the QoS requirements.

Discussion: on the point "Adapt a QoS policy framework for UMTS", it was explained that the work is still going on at IETF and further work at S2 will depend on it.

S2K00-009, source S2: TR 23.821 v.0.1.0

The content of this TR is summarised in the presentation in S2K00-002.

It should be raised in version 1 (50 % stable) for the SA plenary meeting in March. It includes all the changes approved the week before at the last SA2 meeting.

S2K00-004rev1, source T2 chairman: *"All-IP" Impact on T2 – some initial thoughts*

This document is an initial analysis of the impacts on T2 work of the All-IP architecture. The analysis is made for each sub-working group of T2.

On SWG1 (MExE), MExE is independent of the bearers used, however it will need to select and monitor QoS on all types of bearers.

On SWG2 (Terminal Interfaces), again they should be independent of the bearers, however e.g. the AT commands might be modified to ensure that the additional bearers can be controlled.

On SWG3 (Messaging), it should be considered whether it is necessary to support SMS over the packet network if MMS is supported by all packet networks and terminals. MMS will provide all the capabilities of SMS and more so there is no need to provide SMS over the packet network.

On SWG4 (SAT/MEXE/CAMEL interworking), T2 has not started the work in this area.

On SWG5 (Multi-mode Terminals), the implications the all-IP network have on the performance of a multi-mode terminal (e.g. for mobility management, call server access etc.) and the performance of handover of services in both CS and PS domain should be further studied.

And on SWG6 (Terminal Capabilities), The terminal capabilities group has produced several reports, on electrical safety, SAR, etc. These probably don't need to be updated to reflect the All IP network. However the Terminal Capabilities Requirements Report (21.904) will need to be updated with additional information from other working groups and other SWGs within T2.

Discussion: N1 and other WGs working on GSM/UMTS and R99/R00 interworking issues should be involved in particular in the multimode discussions.

S2K00-008, source S1 ad-hoc: *TR 22.976 version 0.5.0*
An ad-hoc group of S1 has just started a new TR (22.976 version 0.5.0) on "Study on PS domain services and capabilities.". This initial draft has not yet been approved by S1.
This document will describe the High level vision, the Applicability of existing toolkits (MExE, Camel, OSA, etc), the new service capabilities and end user benefits, the case study of realisation of some services, the evaluation of what does and does not need to be standardised by 3GPP and the interoperability requirements
Discussion: It was stressed that the services presently supported by CS should not be degraded when migrating to PS. Also the end user should not be aware of whether the service is supported via CS or PS.
The terminology should be consistent between the groups: e.g. PS domain seems not to have the same meaning at S1 and S2.

A set of questions were circulating on the LAN. They are reported in the tdoc S2K00-010.
S2k00-010, source Ericsson: *questions circulating on the LAN on R00*
Ericsson collected the questions circulating in the chat of the LAN and provided this tdoc. The questions are answered one by one in the following.
Question 1: *What combinations of R'00 networks can exist:*
CS Only; PS Only; IM Only; CS + PS; CS + IM; PS + IM; CS + PS + IM ?
Answer: From S2 point of view, IM only is not possible (it requires PS), and CS+IM is not possible neither (it requires PS). All the other configurations should be possible.
In S1, the need for the IM domain has not been established.
It was commented that there should be one "preferred solution", i.e. one single target architecture, and not up to seven possible configurations.

Question 2: *How does the CS R'00 network differ from the CS R'99 network?*
Answer: The CS feature which are not in R99 will be the "classical improvements" as it was between two GSM releases (e.g. improvements of Camel, etc...)

Question 3: *How does the PS R'00 network differ from the PS R'99 network?*
Answer: According to Nokia, in particular the evolution of the QoS part needs to be further studied.
According to Ericsson, the ability to support a great volume of voice over IP service is one of the key new feature.

Question 4: *Backwards Compatibility (USIMs)*
Are R'99 USIMs supported in a R'00 terminal
Are R'99 USIMs supported in a R'00 network (CS+IM or PS + IM)
Are R'00 USIMs supported in a R'99 terminal
Are R'00 USIMs supported in a R'99 network (CS or PS)
Answer: the requirements are not clear at this point. T2 chairman mentioned it should be a S1 issue.

Question 5: *Backwards Compatibility (terminals)*
Are R'99 terminals (CS or PS) supported in a R'00 (CS+IM or PS+IM) network?
Are R'00 terminals supported in a R'99 network
Answer: the question should be extended to earlier releases (e.g. is a R97 terminal supported on a R00 network). This question should be answered by S1 as soon as possible (no answer yet). This should not prohibit other groups to give their point of view to S1.
In 23.922 developed by S2, there is an assumption that the All-IP network shall support the R99 terminals.
Nokia stressed that the risk of fragmentation of the market has to be considered and avoided as much as possible (problems of compatibility between R98 and R99 GSM+UMTS, between R99 and R00).

Question 6: *Backwards Compatibility (services)*
Are R'99 services (CS or PS) supported in a R'00 (CS+IM or PS + IM) network?

Are R'00 services supported in a R'99 network
Answer: For the first one, yes, the R99 services will be supported in a R00 network. They might differ in the way they are supported.
For the second one, the question should be clarified. E.g. Camel phase 4 will provide services supported in R99 but the mechanism itself is obviously not provided by R99.

Question 7: *Roaming*
Can a R'00 terminal roam into a R'99 network
Can a R'99 terminal roam into a R'00 network
Answer: S1 has no answer yet.

Question 8: *Handover*
Can a R'00 terminal handover into a R'99 network
Can a R'99 terminal handover into a R'00 network
Answer: S1 has no answer yet. It raises a lot of related issues (e.g. the fallback mechanisms need to be defined).
In S2 documents (23.821 in particular), there is an assumption that the HO from R00 to R99 has to be supported for PS domain but there is no equivalent statement for the CS domain.

S2K00-007, source GSM North America: *LS on requirements on R00 architecture*
The GSM-NA provide here their opinion on an All-IP network. According to them, the standards process should be driven by service requirements, the introduction of new technologies should improve the customer's service experience (i.e. should not impose a reduction in the service set available or a reduction in the quality of service), and the introduction of All-IP networks should be accomplished in a manner allowing for a smooth integration with existing (i.e. GPRS and circuit switched) technologies to provide a clear and smooth evolution path.
They provide some requirements on the All-IP networks, which shall mainly provide backwards compatibility with the services offered by the Release 99 standard (including basic telecommunication services, supplementary services, and operator specific services), enable provision of services with the same (or greater) quality of service as GPRS and circuit switched services, and the enabling mechanisms (transport technology, etc.) should be transparent to the customer. They shall finally support roaming with non-All-IP networks (including handover / cell re-selection).
Discussion: Once more, it should be clarified that backward compatibility not only with R99 but also with pre-R99 has to be supported.
The methodology of defining R00 at 3GPP was kindly criticised by N1 chairman. E.g. the requirements should definitely be defined before the architecture is developed. Now it seems that the draft from S1 is much less stable that the architecture document from S2.
NTT found the requirements on security ambiguous: some new algorithms have already been defined by 3GPP for UMTS (e.g. mutual authentication). A delegate from GSM NA explained that the LS should not be interpreted as a request from GSM NA for the support of a particular ciphering algorithm.
It was concluded that these requirements have to be considered and should be submitted again to the SA Work Shop on requirements.

S2k00-005, source S2: *LS on Nc, Nb and Mc reference points in R00 architecture.*
With this LS, S2 asks the CN R00 Ad Hoc to initiate the work on the three following reference points: the Mc Reference Point (between MGCF and MGW), the Nc Reference Point (between MSC Server and GMSC Server) and the Nb Reference Point (between two MGWs).
Discussion: It was commented that stage 2 has not yet been completed by SA2, so it might be difficult for CN to start the work in these conditions.
BT stressed that the functionality provided by the involved entities have to be clearly specified before the work can efficiently take place, in particular the role of the MGW (and its links with legacy network e.g. for the support of multimedia over IP) and of the MSC server (and its links with the signalling and bearer control).
A lot of work might also be needed on the CSCF side, and not only on the MSC server as stated by this LS.

A joint meeting between CN and SA2 was proposed, to take place once the S2 material is more stable. The date of April 11th to 13th was proposed to have S2 and N1 discussing together roaming and CC related issues. It was also proposed to have an other joint TSG-CN/SA2 R00 Work Shop established as soon as possible after the TSG#7 plenaries.
No final conclusion of future dates for a joint meeting was finally established. TSG-CN Vice Chairman and SA2 Chairman promissed to propose a date for the next meeting.

(TSG-CN Vice-Chairman and SA2 Chairman met after the meeting and decided to propose the following: TSG-SA2 and TSG-N1 drafting session on roaming and CC related issues on April 10th –12th (until noon) and TSG-CN and SA2 joint R00 Work shop on April 12th (from noon) – 13th both in Helsinki, Finland.)

3 Closing of the meeting

This joint meeting was seen as being very useful by all the participants. The S2 chairman thanked all the participants for their very positive attitude and willingness to progress efficiently the issue. He thanked again the host and the MCC support.

Annex 1. Tdocs list

Tdoc #	Source	Title
S2k00-001	TSG CN chairman	Agenda
S2k00-002	Ericsson + Nokia	Architecture presentation
S2k00-003	S2 chairman	general presentation of SA2 work on R00
S2k00-004	T2	T2 presentation
S2k00-005	S2	LS from S2
S2k00-006	Nokia + S2	Presentation of QoS aspects for R99 and R00
S2k00-007	GSM North America	LS from GSM NA
S2k00-008	S1 ad-hoc	TR 22.976 version 0.5.0
S2k00-009	S2	TR 23.821 v.0.1.0
S2k00-010	Ericsson	Questions from the LAN

Annex 2. Participant list
To be provided latter

3GPP TSG_CN (et.al.) **_S2K0001_**

Puerto Vallarta, Mexico
13-15 December 1999

Source: **Chairman TSG_CN**
harald.dettner@icn.siemens.de

Title: **Draft agenda**

Document for: **Approval**

Meeting style: Workshop / Kick off / Tutorial / "informal"

Start of Meeting: 09:00 local time

Monday 31. January 2000
Joined workshop/tutorial with S2 and T2

1. Introduction / Opening business
2. Presentation of S2 (overall) material
3. Presentation of S2 material relating to CN (plus questions and discussion)
4. Presentation of S2 material relating to T2 (plus questions and discussion)
(if (3) and (4) can already now be separated)
5. General Discussion
   5.1 Technical Aspects
   5.2 "Management" Aspects
   (worksplit, timeframe, co-operation with others, etc.....)
6. Wrap up

Tuesday/Wednesday 01/02. February 2000
CN ad-hoc on IP issues

1. Introduction / opening business

Technical part:

2. Additional material (beside S2) concerning IP-based- (Core-) Networks
   2.1 Presentations & Discussions
3. Technical assessment of presented material
   3.1 "Maturity" of guiding archictural material
   3.2 Delta compared to existing 3GPP specification material (R'99)
   3.3 Compatibility / Optionality aspects
   3.4 ....
4. Miscellaneous

"Management Part"

5. Timeframes
6. Worksplit / Who does what
   6.1 Between TSG's
   6.2 Within TSG_CN
7. Wrap up
8. Next steps until TSG#7
9. Any other business
10. Close of meeting

Release 2000 Architecture

Major Steps in Release 2000

- Two major steps have been identified for R00
  - Call/bearer separation for the CS domain
    - The R99 CS domain architecture evolves by separation of transport and control towards an bearer independent CS domain architecture allowing for IP, and other transport, means internal in the PLMN. (The terminals need no IP capabilities for using services of the CS domain).

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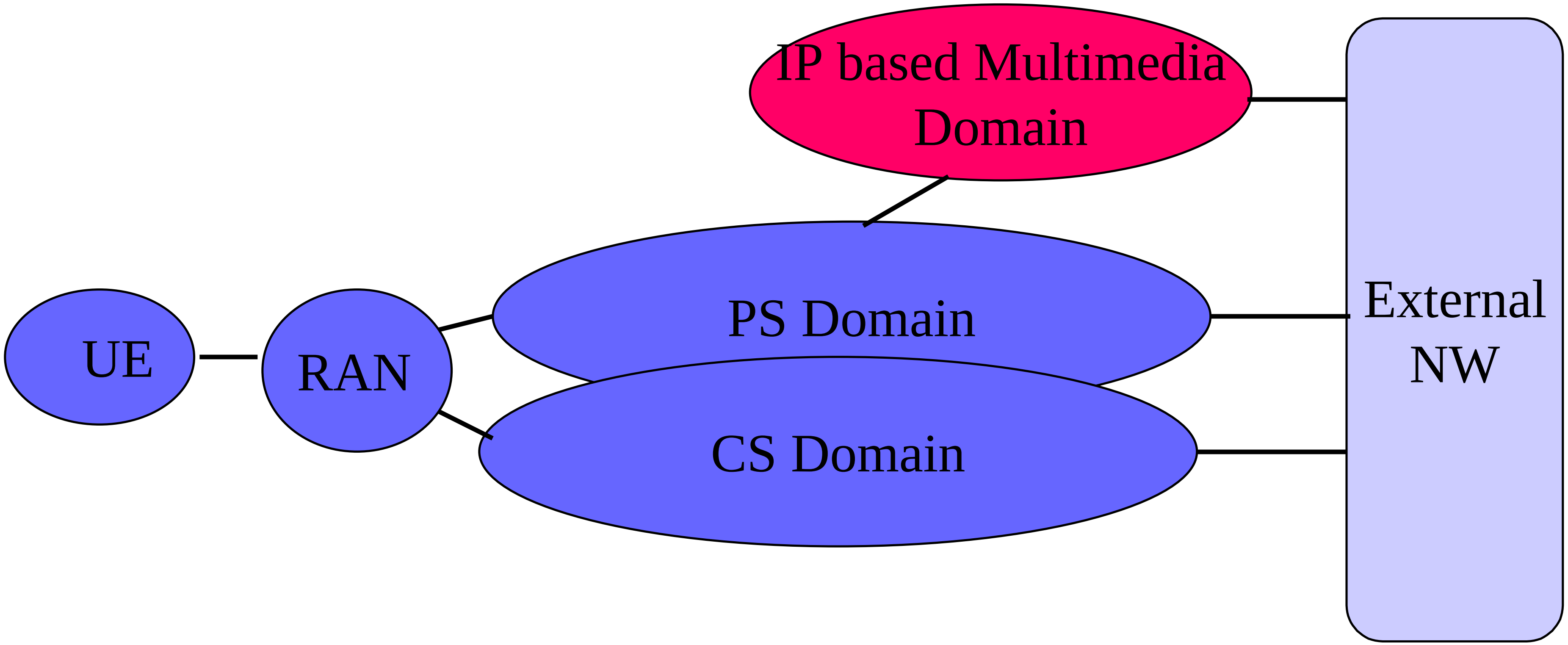
Major Steps in Release 2000

- Addition of IP based Multimedia services as an overlay to the PS domain
  - An IP based multimedia (IM) service control architecture is introduced into the R00 architecture in parallel to the CS domain architecture. The multimedia service control architecture offers services similar, but not the same, to that of the CS domain (e.g. pure voice calls) and in addition services comprising multiple different bearers. The separation of transport and control is inherent. This approach evolves IP based multimedia control standards to support the application services of the mobile network. The service control architecture uses the IP bearer services of the PS domain between the terminal and the network for IP transport. The terminals have to support IP bearer of the PS domain to use IM domain services. The IM domain and the PS domain are architecturally separated allowing any other IP access network.

Domain Descriptions



Domain Descriptions (Preliminary)

- **CS domain**
  - Comprises all core network functionality for provision of bearer- and teleservices in a circuit oriented manner. It includes the functions for the call control, related supplementary services, application services and mobility support. Maintaining calls while terminals change the location is handover functionality of the CS domain UMTS specific call control. Transport and control of the CS domain network are separated to enable service provision by different means of transport resources (ATM, IP, STM, ...) for better transport resource efficiency and convergence with the PS domain transport. An implementation option in the CS domain is the combination of transport and control in one network entity comparable to R99 MSCs. The main new characteristics of the R00 CS domain compared with the R99 CS domain is the flexibility for PLMN internal transport means, that allows for transport based on IP. Between the terminals and the network the protocols are the same as for R99 to use the services offered by the CS domain. This means for example there is no need for IP enabled terminals if IP is the transport resource within the network.

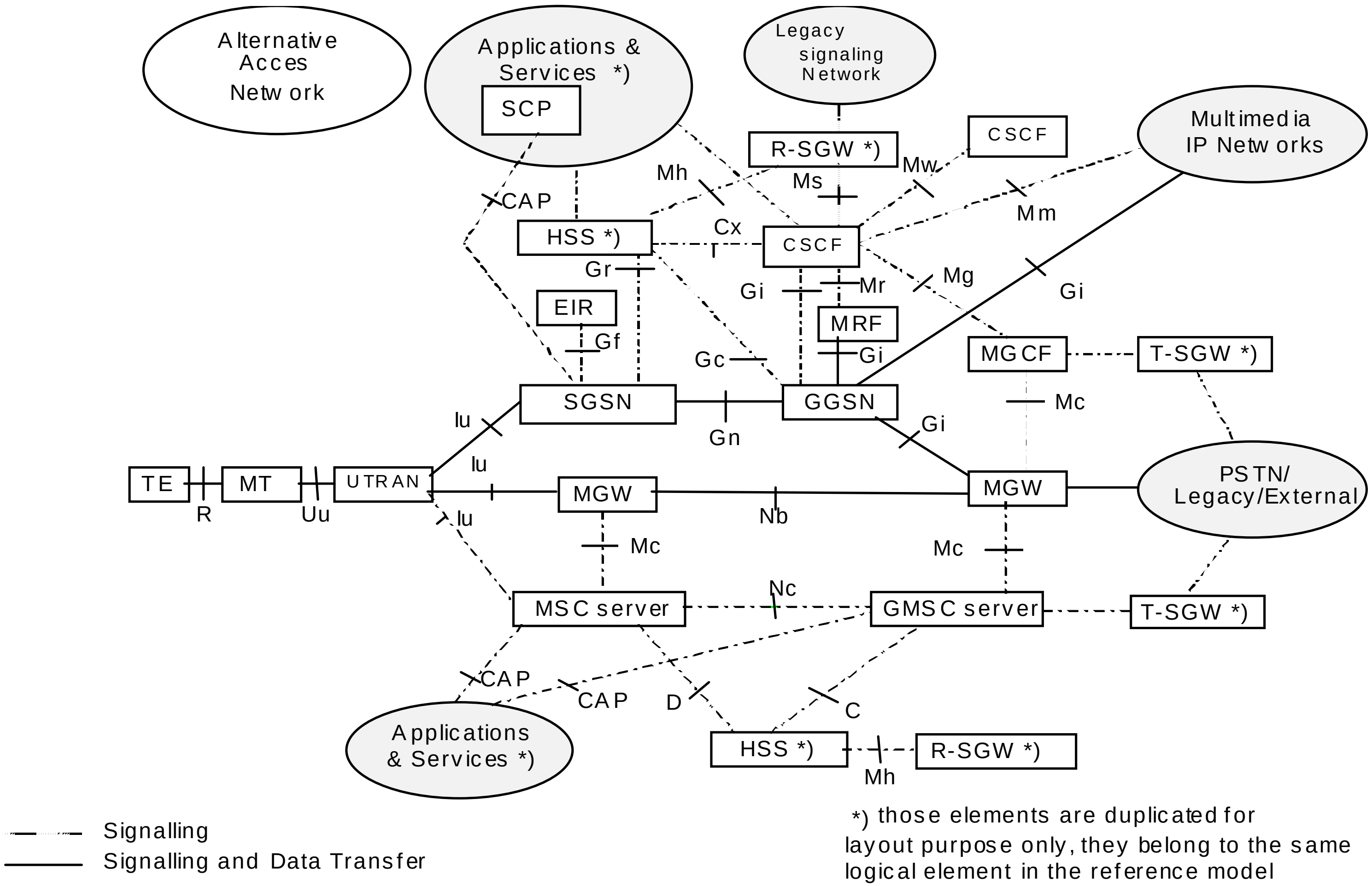
Domain Descriptions (Preliminary)

- **PS domain**
  - Comprises all core network functionality for provision of bearer services in a packet oriented manner. It includes the functionality for bearer service control, application services support and mobility support. The bearer service is maintained while terminals change the location by the bearer service control handover functionality.

Domain Descriptions (Preliminary)

- **IP Multimedia domain (IM)**
  - **enables new services with multiple media components per call based on non-UMTS IP multimedia call control standards (e.g. H.323, SIP). Also, services comparable to that of the CS domain can be offered, e.g. pure voice calls. Terminals using the services of the IM domain have to support the IP bearer services of the PS domain as it provides the IP transport between the terminal and the network. The PS domain bearer service offers the handover functionality for maintaining the service while the terminal changes the location. The separation of IP bearer provided by the PS domain and the call/service control provided by the IM domain results in an IM domain, that is flexible to operate on any other IP access system, e.g. wired networks. The IM domain provides also supplementary services related to the multimedia call control and supports application services.**

Release 2000 Architecture



Release 2000 Architecture (notes)

- The architecture will evolve as the key issues (e.g. roaming, addressing and routing) are resolved.
- A (G)MSC Server and associated MGW can be implemented as a single node as with the (G)MSC in R99
- GERAN is not included in the figure since the issue of GERAN and related interfaces, i.e., Iu_ps, Gb, A are currently being handled in SMG2 and SMG12.

Key Elements

- Call State Control Function (CSCF)
  - Call set-up/termination and state/event management
  - Address Analysis, translation, modification if required, address portability, mapping of alias addresses
  - provide service trigger mechanisms (service capabilities features) towards Application & services network (VHE/OSA)
  - Interacts with HSS in the home domain to receive profile information for the user and
  - Interact with MRF in order to support multi-party and other services

Key Elements

- Home Subscriber Server (HSS)
  – The master database for a given user. It is responsible for keeping a master list of features and services associated with a user, and for tracking of location of its users. It is a superset of the HLR functionality.The HSS holds a subscription profile which identifies for a given user for example:
    - user identities
    - subscribed services and profiles
    - service specific information
    - mobility management information
    - authorisation information

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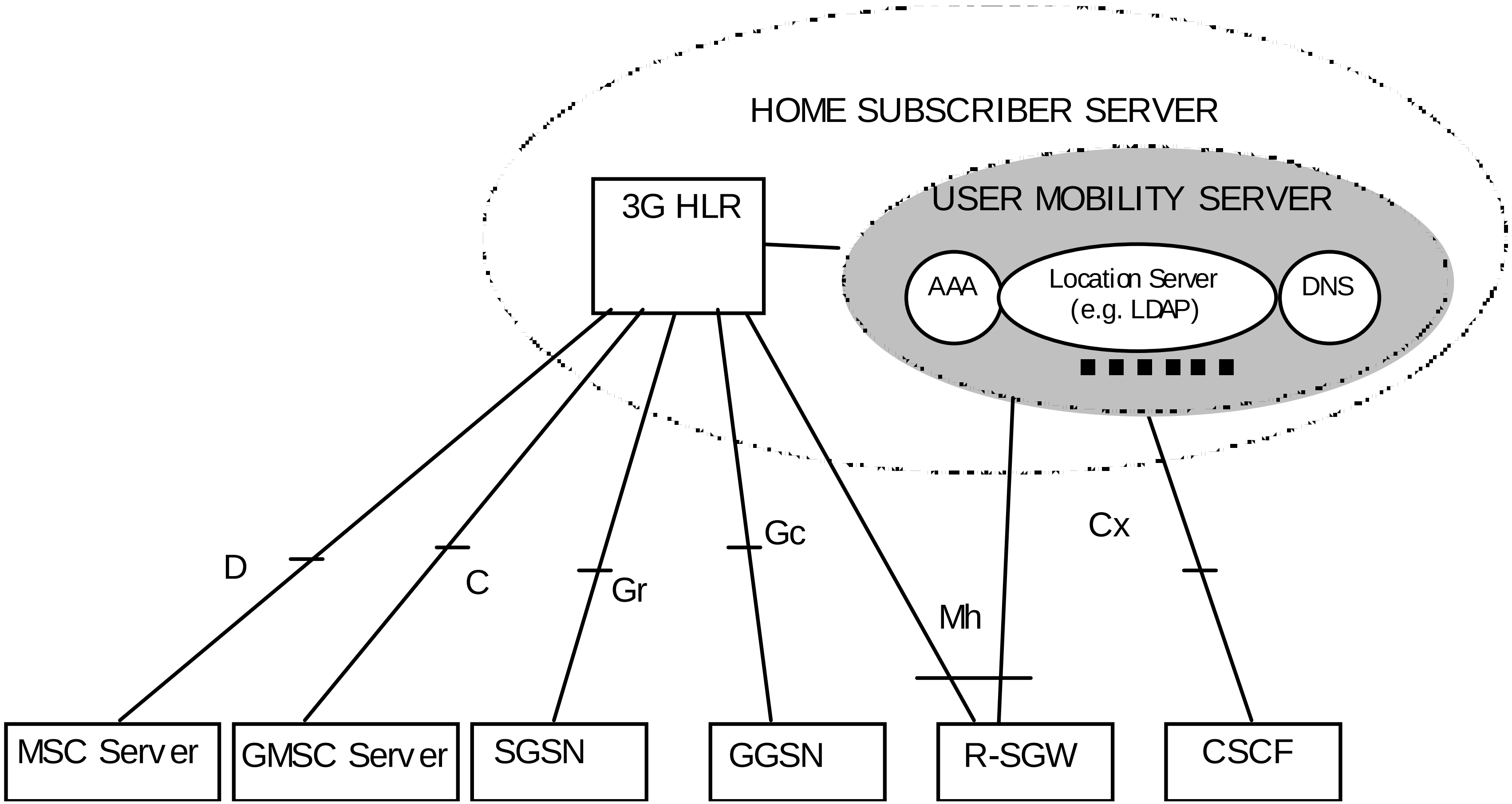
Key Elements



Key Elements

- Media Gateway Control Function (MGCF)
  - Performs protocol conversion between the Legacy (e.g. ISUP, R1/R2 etc.) and the IM domain call control protocols.
  - Controls the parts of the call state that pertain to connection control for media channels in a MGW.
  - MGCF selects the CSCF depending on the routing number for incoming calls from legacy networks.

Key Elements

- Transport and Roaming Signalling Gateway Function (T-SGW, R-SGW)
  - The R-SGW and T-SGW performs signalling conversion at transport level (e.g. Sigtran SCTP/IP versus SS7 MTP)

Key Elements

- Multimedia Resource Function (MRF)
  - Performs multiparty call and multi media conferencing functions. MRF would have the same functions of an MCU in an H.323 network.
  - Is responsible for bearer control in case of multiparty/multimedia conference.
  - May communicate with CSCF for service validation for multiparty/multimedia sessions.

Key Elements

- Media Gateway Function (MGW)
  - This component is PSTN/PLMN transport termination point for a defined network and interfaces UTRAN with the core network over Iu.
  - A MGW may terminate bearer channels from a switched circuit network 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).
    - Interacts with MGCF, MSC server and GMSC server for resource control.
    - Owns and handles resources such as codecs, echo cancellers etc.

Key Elements

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

Key Elements

- Gateway MSC Server
  - The GMSC server mainly comprises the call control and mobility control parts of a GSM/UMTS GMSC.
- (G)MSC
  - A (G)MSC server and a MGW make up the full functionality of a (G)MSC as defined in 23.002 (G)MSC

Key Reference Points

- 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.

Key Reference Points

- Gm Reference Point (CSCF – UE)
  - This interface is to allow UE to communicate with the CSCF to:
    - Register with a CSCF
    - Call origination and termination
    - Supplementary services control

Key Reference Points

- Mc Reference Point (MGCF – MGW)
  - The Mc reference point describes the interfaces between the MGCF and MGW, and between the (G)MSC Server and MGW. It has the following properties:
    - full compliance with the H.248 standard
    - flexible connection handling which allows support of different call models and different media processing purposes
    - 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

Key Reference Points

- Mw Reference Point (CSCF – CSCF)
  - The interface allows one CSCF (e.g. home CSCF) to relay the call request to another CSCF (e.g. serving CSCF).

Key Reference Points

- Nc Reference Point (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 R'00 architecture different options for signalling transport on Nc shall be possible including IP.

Key Reference Points

- Nb Reference Point (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. In the R00 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.

Key Reference Points

MAP/CAP interfaces

MAP/CAP		
TCAP		
SCCP		
M3UA	MTP-3B	MTP-3
SCTP (1)	SAAL	
IP (2)	ATM(2)	STM (2)

Note: The protocols do not correspond to the same OSI layer. They are drawn on the same height as they are "transport alternatives".

Key Reference Points

- Iu Reference Point
  - This is the reference point between UTRAN and the R00 core network. This reference point is realised by one or more interfaces:
    - Between UTRAN and SGSN, transport of user data is IP based.
    - Between UTRAN and SGSN, transport of signalling is based on IP or SS#7.
    - Between UTRAN and MGW, transport of user data is based on different technologies (e.g., IP, AAL2), and includes the relevant bearer control protocol in the interface.
    - Between UTRAN and MSC server, transport of signalling is based on IP or SS#7.

Key Reference Points

- Iu Reference Point
  - When the Iu_cs is ATM based, then the protocols used can be based on R99 protocols or an evolved version.
  - When Iu_cs is IP based, new IP transport related protocols need to be added as part of the Iu protocols. It shall be possible to have R99 Iu interface with MSCs compliant to R99 specifications in the network.
  - It shall be possible to have a R99 CS domain with R99 Iu_cs reference point coexisting with a R00 Iu reference point.

Key Issues

- Most urgent issues to be solved for the R00 architecture
  - Domain Definitions
  - Roaming
    - What is the functionality split between the visiting and home networks?
      - Where to support the services/applications
      - Into/from Legacy Networks
      - Where to support the call control (When roaming outside the home network, does the CC run from the MS to the visited network [roamed to] and/or transparently through the visited network to the home network?)
      - Where to locate the bearer/resources e.g local routing, is the MRF used in the visited/home network or both.

Key Reference Points

- Addressing, Identifiers and Routing
  - What are the identities of the subscriber and the aliases of the terminal (Addressing Principles)
    - Identifiers external (e.g. MSISDN, Ip address, aliases, domain name, e-mail)
    - Identifiers inside the network multiple levels (e.g. bearer, application)
  - Where is the IP address allocated?
  - What is the scope of the IP address (which parts of the network is it relevant within)?

Key Reference Points

- General
  - Standard non-3GPP CC and multi media mechanisms, or mobile enhancements to a non 3GPP CC and multi media mechanism, or a 3GPP specific CC and multi media mechanism.

S2K 0003

3GPP TSG-SA2 Release 2000 Work in General

TSG-CN R00 Work Shop Tutorials

31.01.2000

Teuvo Järvelä

TSG-SA2 Chairman

3GPP Project Co-ordination

- S1 defines the features and services required
- S2 defines the architecture for the features and :
  - divides the features into building blocks.
  - forwards the building blocks to the relevant TSGs for the detailed work.
  - Interactively work with TSGs/WGs for common understanding.

3GPP Project scheduling

- S1 sets a target
- S2:
  - performs a first technical review and comment on the target.
  - indicates target for time schedule with allocation of the defined building blocks.
  - Aligns targets with TSG and WG comments
- S1 and S2 involve S3 to ensure security

3GPP Project Co-ordination and SA2

- Inter-Group Co-ordination ad-hocs (IGCs):
  - Bearer Services and QoS (Oscar-Lopez Torres, T-Mobil)
  - GSM/UMTS Interoperation and Mobility Management (Francois Courau, Alcatel)
  - LCS and CBS (Jan Káll, Nokia and Martin Guntermann, Mannesmann)
  - Packet Architecture and Circuit Architecture (Ulrich Dropmann, Siemens)
  - Security (Chris Pudney, Vodafone)
  - Services and Service platforms (Christophe Gourraud, Ericsson)
- S2's role is to Co-operate with the TSGs/WGs

R00 Architecture work

- Creation of a new TR on Release 2000 Architecture (TR 23.821)
  - Not intended to be published.
  - SA2 R00 Architecture Related decisions will collected into this document and moved to relevant documents after completion of R99 work (CRs will be created between March and May 2000)
- Reference Model and Key Issue list agreed
- Schedule for R00
  - S2 R00 work 50% or more mature for SA #7 (v.1.0.0)
  - S2 Project Plan work for R00 (target date: SA #7)
  - S2 R00 work 80% or more mature for SA #8 (v.3.0.0)
  - S2 R00 work finalised by SA #9

SA2 Meeting Dates

- TSG-S2#11 January 24-28, 2000. AT&T, Ericsson, Nokia
- TSG-S2#12 March 6-9, 2000 - Tokyo, Japan, NTT and NTT DoCoMo
  - Dates reserved for drafting groups on Key Issues
    - April 11-13
    - April 25-27
    - May 9-11
- TSG-S2#13 May 22-26, 2000 - Berlin, Germany, Siemens
- TSG-S2#14 September 4-8, 2000 - Bristol, UK
- TSG-S2#15 November 13-17, 2000 - host needed

3GPP Joint CN/S2/T2 on R00
Puerto Vallarta, Mexico, 31st Jan 2000

CN/S2/T2#1 (00)0004

Agenda Item:

Source: T2 Chairman

Title: "All-IP" Impact on T2 – some initial thoughts

Document for: Discussion

The work in T2 is generally focused on applications and APIs for the terminal. In principle many of the applications/APIs will work independently of whether used over a circuit switched or packet switched link. However it is worthwhile looking at the detail of the T2 work to determine whether there are any impacts. The following is an initial analysis and comments are invited:

SWG1 – MEXE

MEXE specifies the terminal API and security environment for downloaded services and applications. This should be independent of the bearers used, however MEXE will need to select and monitor QOS on any bearer provided. It seems likely that MExE will be a significant solution for the provision of current look and feel of circuit switched supplementary services in a packet domain where operators move to All IP based look and feel for supplementary services. Additionally, MExE also provides the service capability to create all-IP variants of existing SS (e.g. multimedia call forwarding) and completely new 3G services.

MEXE may also need to know the type of network connected and for example what call servers are available in the network. This type of information should already be available in the terminal and it is just a question of making sure that the information is available to MEXE applications.

SWG2 – Terminal Interfaces

The terminal interfaces should be independent of the bearers used, however again it may be necessary to make some additions to for example the AT commands to ensure that the additional bearers can be controlled, or options on the handset (e.g. user's preference for call server location).

SWG3 – Messaging

This group is looking into Cell Broadcast, SMS and Multimedia Messaging (MMS). Cell Broadcast should be available via the packet channels in a system only providing packet channels (really an S1/S2 issue). From the perspective of moving forwards it should be considered whether it is necessary to support SMS over the packet network if MMS is supported by all packet networks and terminals. MMS will provide all the capabilities of SMS and more so there is no need to provide SMS over the packet network. Of course, interworking to SMS on other networks is needed.

SWG4 - SAT/MEXE/CAMEL interworking

T2 has been unable to start the work in this area. This has partially been picked up by the work in T3 on SAT/MEXE interworking and also the OSA work in CN. However it is not clear whether someone has an overall picture from the application (T2-area) perspective.

SWG5 – Multi-mode Terminals

The multi-mode terminals group has studied work going on in other working groups of 3GPP and has performed some analysis of terminals which have to work in different modes (e.g. GSM and R99-CS). The implications the all-IP network have on the performance of a multi-mode terminal (e.g. for mobility management, call server access etc.) and the performance of handover of services in both CS and PS domain could be appropriate subjects for further studies in this group.

SWG6 – Terminal Capabilities

The terminal capabilities group has produced several reports, on electrical safety, SAR etc. These probably don't need to be updated to reflect the All IP network. However the Terminal Capabilities Requirements Report (21.904) will need to be updated with additional information from other working groups and other SWGs within T2.

3GPP Joint TSG_CN, T2, S2 on R00 **S2K00-005**

Puerto Vallarta, Mexico , 31 January 2000

TSG-SA Working Group 2 meeting #11 **TSGS2#11 S2-000317**

Puerto Vallarta, Mexico, 24-28 Jan 2000

Title: LS on Nc Nb and Mc reference points in Release ´00 architecture.

To: 3GPP TSG-CN

CC: 3GPP CN R00 Ad Hoc

Source: 3GPP SA WG2

SA2 has been working on the R'00 reference architecture, and would like to inform the CN R00 Ad Hoc of the progress on three of the interfaces in the architecture. The latest proposed R´00 reference architecture is reproduced below in Figure 1 below, with the reference points further discussed in this document (Mc, Nc and Nb) highlighted. The latest technical report (TR 23.821 v 0.1.0) is included as an attachment.

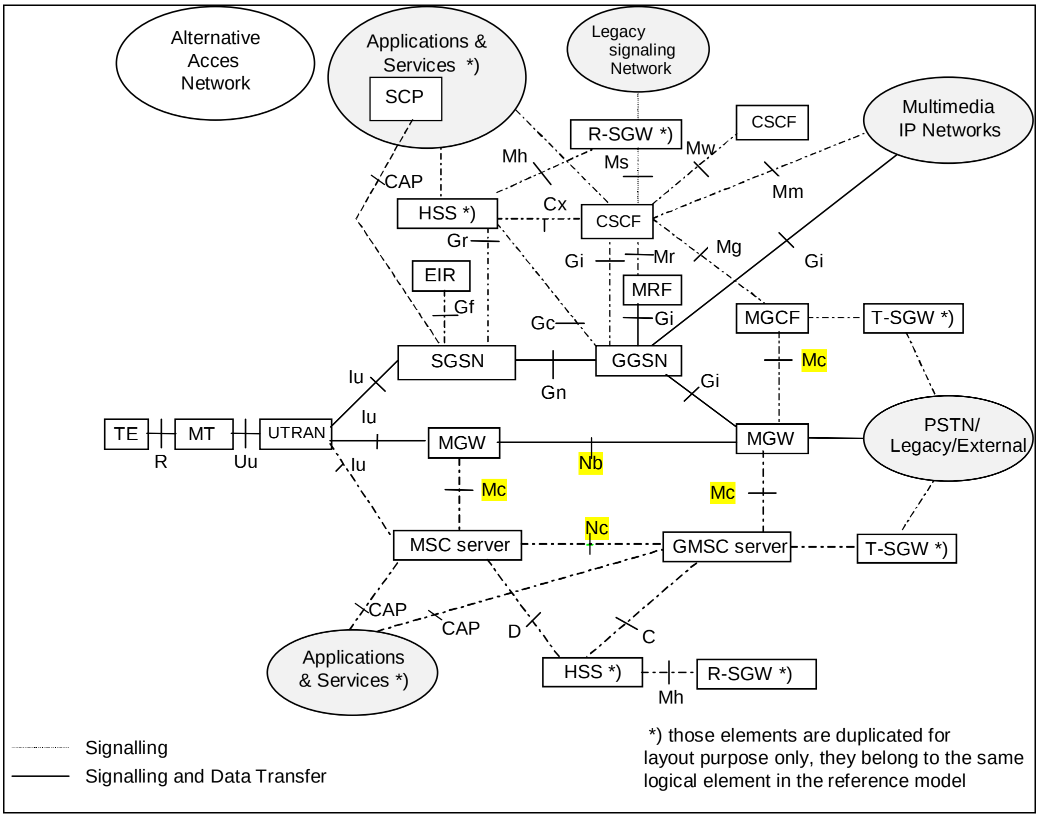


Figure 1 Proposal for R'00 reference archecture

Mc Reference Point (MGCF – MGW)

The Mc reference point is defined as :

> *"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."*

H.248 / Megacop was designed to be applied to a wide range of applications, from large trunking gateway applications, to small IP phone applications. As such, it includes a number of options and is very flexible. Each application which will utilise H.248 is required describe how it will apply H.248/Megacop to its system, by selecting the options which are valid and selecting/developing the appropriate set of "packages".

The nature of the Mc reference point between the MGCF and MGW will be dependant upon the call control (e.g. H.323; SIP; …) used. As it not expected that 3GPP systems will place additional requirements on this interface, the packages developed in the relevant forum (i.e. ITU-T SG-16 for H.323; IETF for SIP;) should be sufficient. It is proposed that 3GPP evaluate the available options for coding, signalling transport, and valid packages and specify which are valid for 3GPP systems on this interface.

The requirements on the Mc reference point between the GMSC server and the MGW are already covered by other industry forums such as IETF, ITU-T SG-16 and ITU-T SG-11 for trunking gateway applications, as such it is expected no additional packages are required to be defined for 3GPP. It is proposed that 3GPP will evaluate the available options for coding, signalling transport, and packages and specify which are valid for 3GPP systems.

In the realisation of the Mc reference point between the MSC server and MGW, additional requirements are placed upon 3GPP systems in order to support :

- Circuit Switched Data
- Iu userplane termination as described as the Iu CS in 25.410 V 3.1.0 "*UTRAN Iu Interface: General Aspects and Principles"* for R´99 and R´00.
- Ensure/verify that H.248 provides support for inter RNC handover.

It is expected that 3GPP specifies a package for H.248/Megacop which will support circuit switched data and the termination of the Iu userplane, and evaluate the available options for coding, signalling transport and valid packages and specify which are valid for 3GPP systems.

Nc Reference Point (MSC Server – GMSC Server)

The Nc reference point is defined as

> *"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 R'00 architecture different options for signalling transport on Nc shall be possible including IP."*

A protocol suitable for the Nc interface is being developed in ITU-T SG-11 for the support of ATM and IP bearers. Capability Set 1 of the protocol, Bearer Independent Call Control, has been completed (determined as Q.1901). Capability Set 2, which will include support for IP bearers and the separation of the MSC/GMSC server and the media gateway, is expected to be determined at the end of 2000.

Specification of the BICC protocol itself is not required within 3GPP.

It is expected that a document similar to "ISDN-Public Land Mobile Network (PLMN) signalling interface" (GSM 09.12) may be required from 3GPP to describe the use of BICC for ATM and IP bearers for the R´00 architecture.

Nb Reference Point (MGW – MGW)

The Nb reference point is defined as:

> *"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 dataIn the 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."*

The protocols suitable for the Nb reference point are being developed in other industry forums, and is dependant upon the bearer technology employed.

In the case of AAL2 transport, the AAL2 userplane protocols as defined in I.363.2 (AAL2), Q.2630.1, and associated recommendations defined in ITU-T, are appropriate. In the case of IP transport, the RFCs for the Userplane transport (RTP), and the recommendations associated with BICC for the establishment of IP bearers are appropriate.

It is expected that a specification covering the application of the Nc interface also describes the application of the Nb reference point for IP and ATM/AAL2 transport.

Conclusion

Based on the information presented in this liaison statement, SA2 requests that R´00 Ad Hoc initiates the indicated specification work required to realise these interfaces.

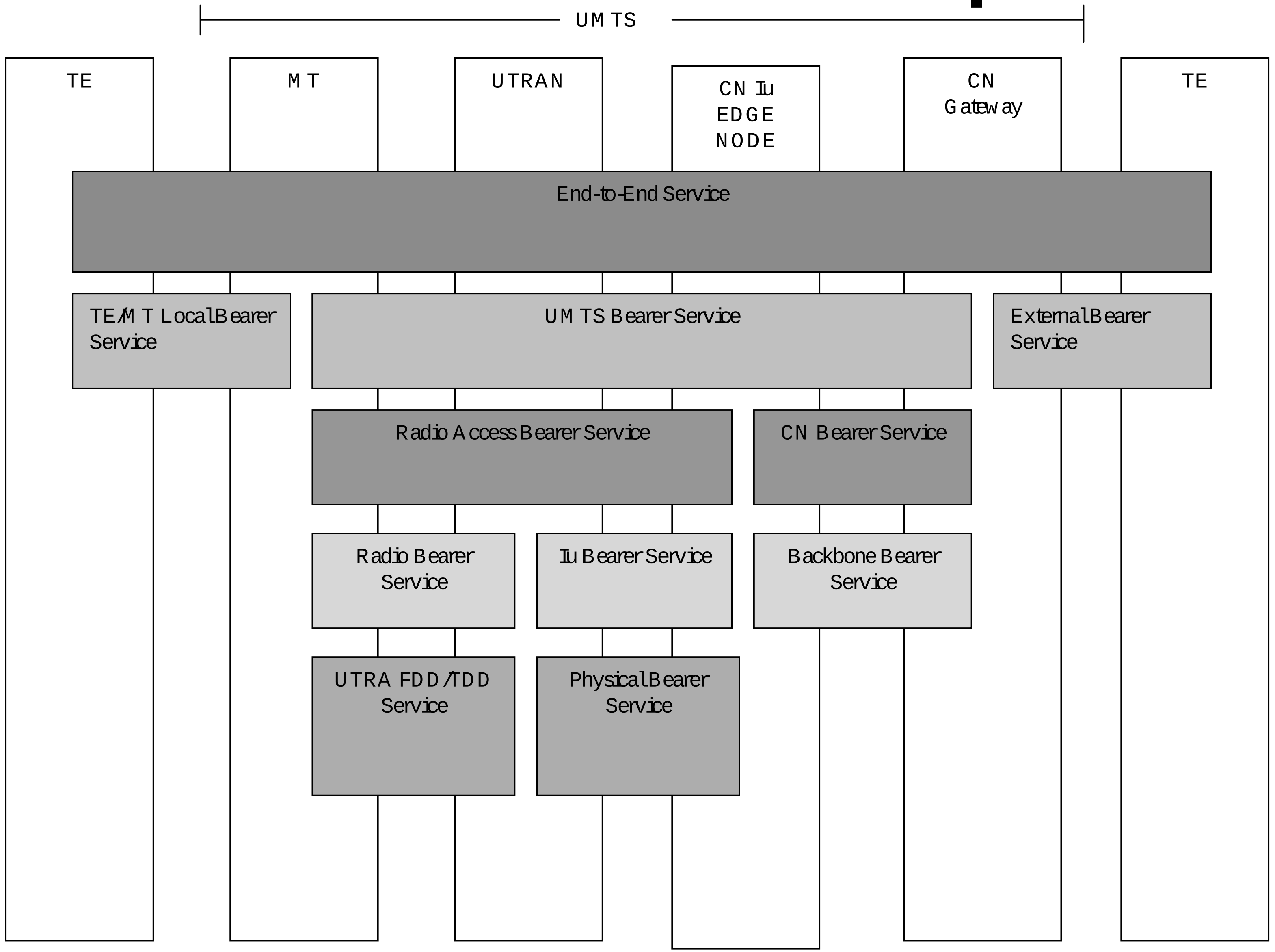
R99 UMTS QoS Concept and Architecture

Presentation by S2

Key Concepts in UMTS R99 QoS

- 3GPP TS 23.107 "QoS Concept and Architecture" v 3.1.0 approved and stable for Release 99
- Multiple PDP contexts with the same PDP address, packet multiplexing at GGSN based on Traffic Flow Templates, UE controlling PDP context activation/modification and TFT's.
- UMTS traffic classes and new QoS parameters (same set of parameters for PS and CS)
- UMTS Bearer Service parameters mapped onto different interfaces and layers
- Some QoS functions identified at each network element: policing, monitoring, packet classification, marking, etc.

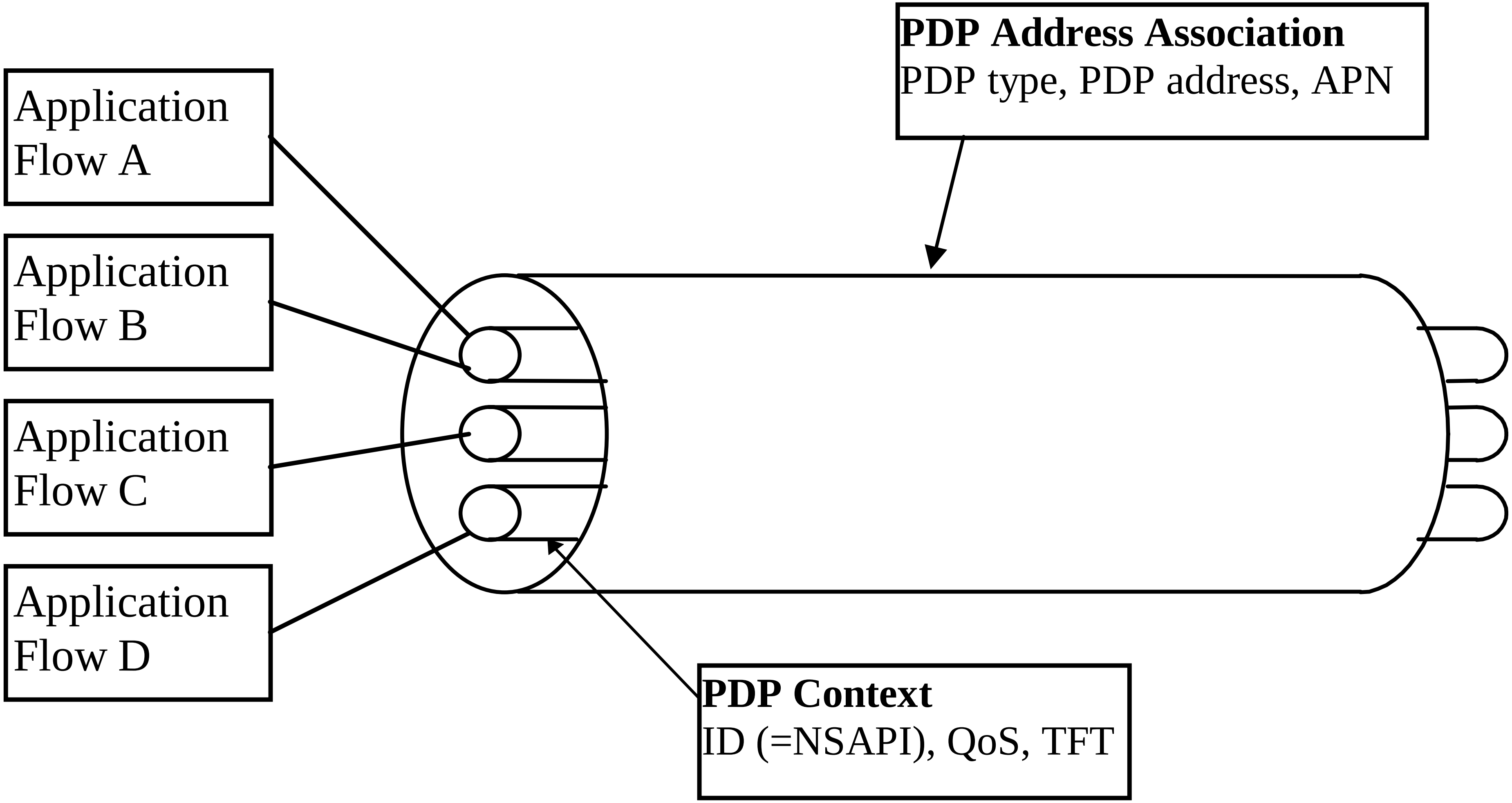
UMTS R99 QoS concept



UMTS Traffic classes

Traffic class	Conversational class conversational RT	Streaming class streaming RT	Interactive class Interactive best effort	Background Background best effort
Fundamental characteristics	• Preserve time relation (variation) between information entities of the stream • Conversational pattern (stringent and low delay)	• Preserve time relation (variation) between information entities of the stream	• Request response pattern • Preserve payload content	• Destination is not expecting the data within a certain time • Preserve payload content
Example of the application	- voice	- streaming video	- Web browsing	- background download of emails

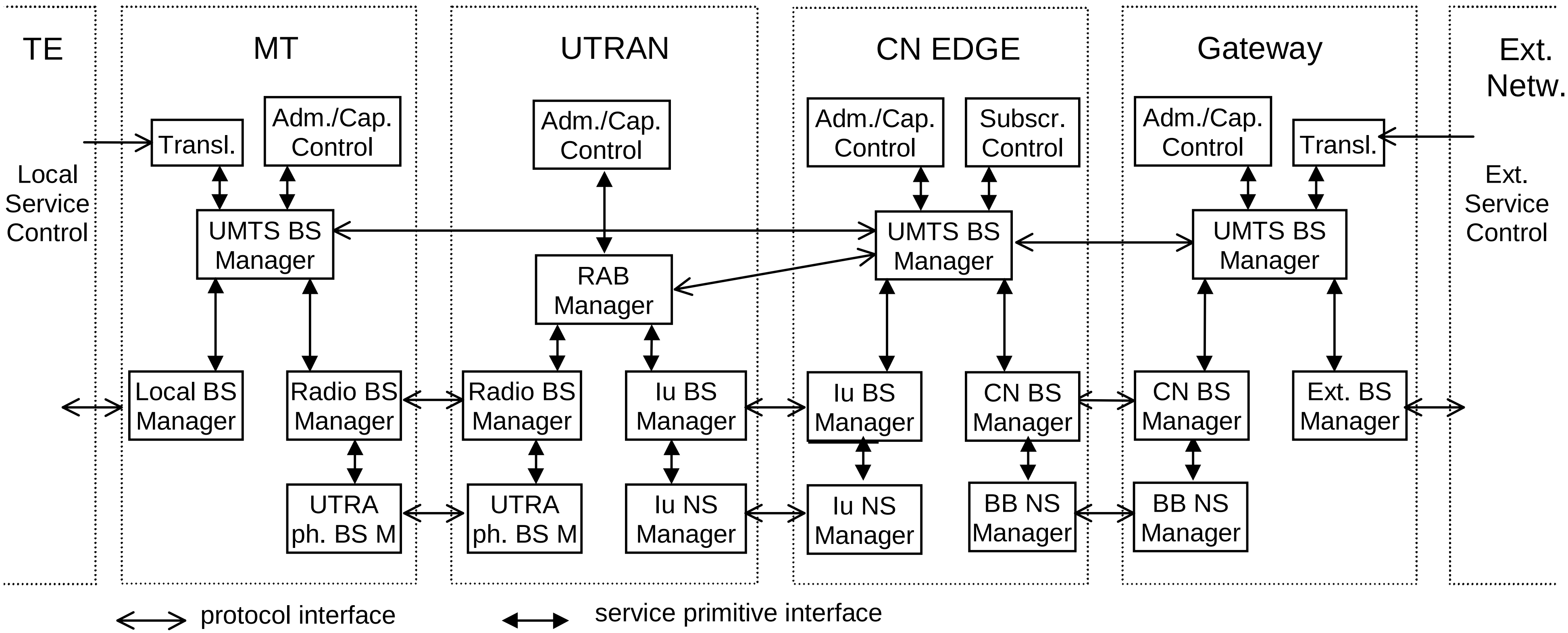
Multiple PDP context concept



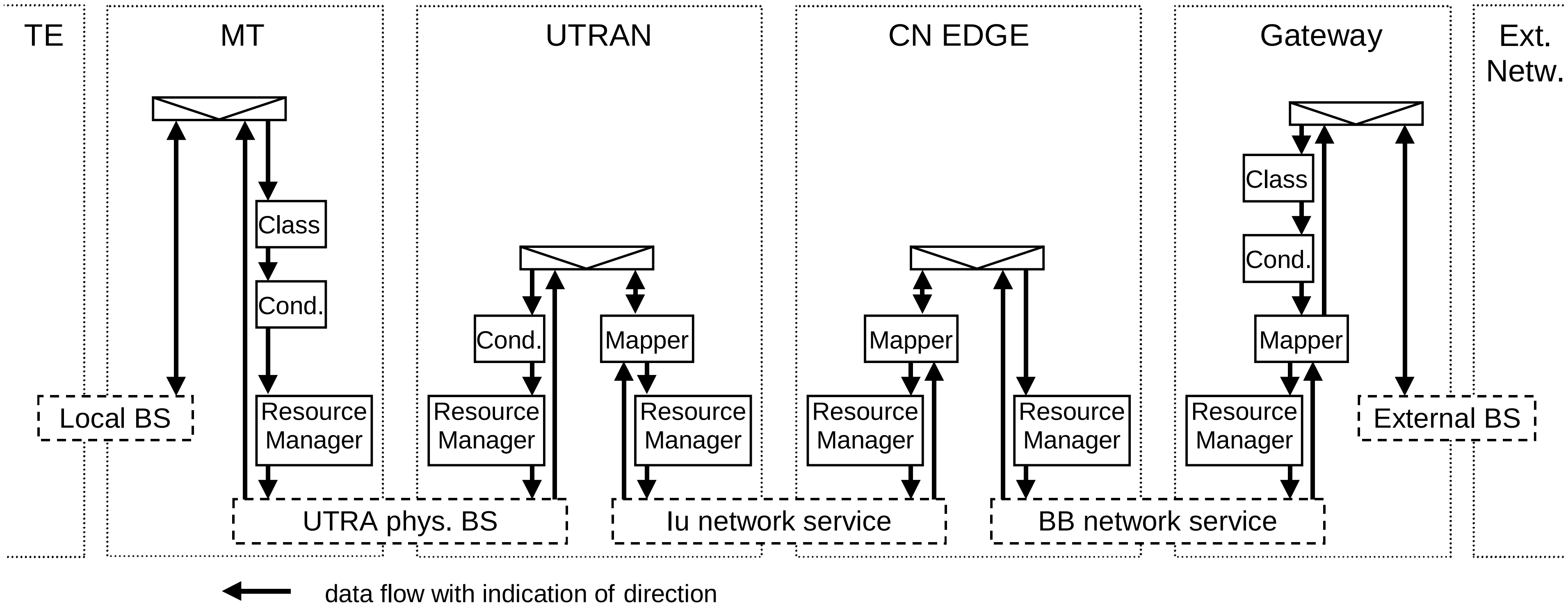
Traffic Flow Templates

- Traffic Flow Template = information available for the GGSN for multiplexing of downlink data packets onto several secondary PDP contexts, i.e. information used to select the right PDP context for a data packet
- TFT consists of independent filters
- Possible filters include:
    - Flow label (only applicable for Ipv6)
    - DS codepoints (Type of Service field in IPv4, Traffic Class in IPv6)
    - Source port
    - Destination port
    - Source address
    - SPI (Security Parameters Index for IPSec)
    - Protocol number
- Also wildcards are allowed

QoS management functions



QoS in the user plane



UMTS Bearer Services QoS parameters

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate	X	X	X	X
Delivery order	X	X	X	X
Maximum SDU size	X	X	X	X
SDU format information	X	X		
SDU error ratio	X	X	X	X
Residual bit error ratio	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/Retention priority	X	X	X	X

UMTS Bearer Service parameter value ranges

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate (kbps)	<2000 (1) (2)	<2000 (1) (2)	< 2000 - overhead (2) (3)	<2000 - overhead (2) (3)
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size (octets)	<1500 (4)	<1500 (4)	<1500 (4)	<1500 (4)
SDU format information	(5)	(5)		
Delivery of erroneous SDUs	Yes/No/- (6)	Yes/No/- (6)	Yes/No/- (6)	Yes/No/- (6)
Residual BER	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4} (7)	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6} (7)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (8) (7)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (8) (7)
SDU error ratio	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (7)	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (7)	10^{-3}, 10^{-4}, 10^{-6} (7)	10^{-3}, 10^{-4}, 10^{-6} (7)
Transfer delay (ms)	100 – maximum value(7)	500 – maximum value (7)		
Guaranteed bit rate (kbps)	<2000 (1) (2)	<2000 (1) (2)		
Traffic handling priority			1,2,3 (9)	
Allocation/Retention priority	1,2,3 (9)	1,2,3 (9)	1,2,3 (9)	1,2,3 (9)

Note: Footnote explanations are given in 23.107 Section 6.5.1

UMTS Bearer Service parameters & mapping

- Application requirements mapped by UE to the UMTS Bearer Service QoS parameters [mapping at UE is implementation specific]
- UMTS Bearer Service QoS parameters mapped to underlying layers
  - Almost the same parameter set at the Radio Access Bearer level, some parameter values need to be adapted
  - Mapping to Iu and Gn interfaces left as an operator's choice but Differentiated Services shall be supported at IP level
  - Differentiated Services specified in IETF
  - Service Level Agreements provide interworking between domains

Key issues in R00

- End to end quality of service negotiations and provision (e.g. delay)
- The applicability of QoS negotiation mechanisms e.g. RSVP signalling
- Adapt a QoS policy framework for UMTS
- Interact between CC/application signalling and QoS negotiation
- What are the Qos requirements to carry application and control signalling (e.g. CC/Multi-media signalling)?
- Ongoing support for inter SGSN change (handover and SRNS relocation), both within GSM and 3G and between GSM and 3G, to satisfy the QoS requirements.

GSMNA Doc 074/00
NaStd00/047R1

GSM North America
The North American Interest Group of the GSM Association

January 14, 2000

To: Alan Cox, Chair, 3GPP SA1
Harald Dettner, Chair, 3GPP CN
Niels Andersen, Chair, 3GPP SA
Tuevo Jarvela, Chair, 3GPP SA2
cc: Asok Chatterjee, Chair, T1P1
Linda Melvin, Director, GSM-NA
Gary Jones, Chair, GSM-NA Standards Working Group
M.V. Thomas, Chair, GSM-NA Services Working Group
Randolph Wohlert, Chair, GSM-NA Location Services SubWorking Group
Philippe Lucas, Chair, GSM Association SERG Group

Dear Chairpersons,

The GSM-NA is interested in the important work taking place within your organization regarding standardization of an All-IP network.

In our opinion:
- The standards process should be driven by service requirements, and
- The carrier community should be an integral part of this process, and
- Introduction of new technologies should improve the customer's service experience (i.e. should not impose a reduction in the service set available or a reduction in the quality of service).
- The introduction of All-IP networks should be accomplished in a manner allowing for a smooth integration with existing (i.e. GPRS and circuit switched) technologies to provide a clear and smooth evolution path.

The following high-level service requirements need to be incorporated for successful deployment of services based on an all-IP network. These should be considered when identifying more specific requirements.

All-IP networks shall

- Provide backwards compatibility with the services offered by the Release 99 standard (including basic telecommunication services, supplementary services, and operator specific services)

  - The set of services available to customers shall be no less than the set of services available to customers obtaining service using existing GPRS and circuit switched technologies.

- Enable provision of services with the same (or greater) quality of service as GPRS and circuit switched services.

  - It shall be possible to offer services over an All-IP network with a quality of service that is no

less than that already experienced by customers of existing GPRS and circuit switched networks.
- The enabling mechanisms (transport technology, etc.) should be transparent to the customer.

- Enable provision of the same (or greater) degree of privacy, security, and authentication as GPRS and circuit switched services.

- Support roaming between All-IP networks and non-All-IP networks (including handover / cell re-selection).

We respectfully request your organization to incorporate these requirements as part of the All-IP Service requirements specification. They may also prove useful as guidelines for more detailed discussions on specific requirements and standards decisions.

Sincerely yours,

[signed copy on file]

Jim Murrell
Chair, GSM North America

3GPP TSG_CN
CN_R00 Ad Hoc Meeting #1, Puerto Vallarta, Mexico
1st – 2nd February 2000.

Tdoc NP-000002

Disclaimer: This TR is an ad-hoc draft, and subject to modification.
The TR has yet to be presented to, or be accepted by, TSG-S1.

3G TR 22.976 V0.5.0 (2000-01)

Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Study on PS domain services and capabilities
(3G TR 22.976 version 0.5.0)

Reference

DTS/TSGSA-0122004U

Keywords

3GPP, SA

3GPP

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Contents

Foreword

This Technical Specification 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 3.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 provides background information, motivations, descriptions of service drivers, and concepts regarding general service requirements and service features of the All-IP network option. The All-IP network option will be developed in a phased approach. The scope of the first phase of the All-IP network option (as part of 3GPP Release 2000) is describedThis TR provides the basis for the detailed Stage 1 specification work.

The focus of the TR is:

- High level vision
  - Potential Service Drivers
- New service capabilities and end user benefits
- Case study of realisation of a some services e.g. CFU
- Evaluation of what does and does not need to be standardised
- Release roadmap
  - Feature List
  - Time of delivery expectations for standards and products (now indented)
- Division of responsibility between S1 and S2, dialogue between the two groups is required. There is a need to set expectations.

This TR has been created to ease the development of the All IP network option, and this document can be used to guide 3GPP in the creation of new specifications and CRs to existing specifications for the realisation of mobile communications services based on the Release 2000 specifications.

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] GSM 01.04: "Digital cellular telecommunications system (Phase 2+); Abbreviations and acronyms".

[2] TR 21.905: "Vocabulary for 3GPP Specifications"

[3] TS 22.101: "Service principles"

[4] TS 22.105: "Services and Service Capabilities"

[5] TS 22.060: "General Packet Radio Service (GPRS) stage 1"

[6] TS 22.003: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Public Land Mobile Network (PLMN)"

Editor's note: Update spec name …

[7] TS 22.004: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; General on supplementary services"

[8] TS 22.121: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; The Virtual Home Environment"

[9] TS 22.057: ”; Mobile Station Application Execution Environment (MExE); Service description, Stage 1”

[10] TS 22.078: ”; Customised Applications for Mobile network Enhanced Logic (CAMEL); Service definition - Stage 1”

[11] GSM 11.14: ”Digital cellular telecommunication system (Phase 2+); Specification of the SIM Application Toolkit for the Subscriber Identity Module - Mobile Equipment; (SIM - ME) interface”

Editor's note: Stage 1 required

[12] *> reference to 22.001…*

[13] TR 21.978: Feasibility Technical Report - CAMEL Control of VoIP Services

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this TR the following definitions apply:

All IP network: an integrated telecommunications network that uses IP for the transport of all user data and signaling

Editor's note: may need this to elaborate from an IP services point of view (i.e. support of IP addresses and IP clients in the terminal)

Circuit switched (CS) domain: the CS domain comprises all network functionality for provision of bearer and teleservices in a circuit orientated manner

Circuit services: the services enabled by the circuit switched domain

Editor's note: CS domain and PS domain may require further elaboration

Emergency call: a mobile originated basic call that terminates at a national or local emergency center. Provision of location information to the emergency center is a mandatory feature in some countries

IP telephony: a voice call that uses IP for transport of all user data and signalling

Multimedia service: multimedia services are services that handle one or more media simultaneously such as speech (e.g. IP telephony), audio, video and data in a synchronised way from the user's point of view. A multimedia service may involve multiple parties, multiple connections, and the addition or deletion of resources within a single communication session

Operator specific service: any service offered to a mobile user that is not standardised by the 3GPP specifications

Packet switched domain: the PS domain comprises all network functionality for provision of bearers in a packet orientated manner

Release 99: 3GPP specified release of complete technical specifications for the definition and development of telecommunication services (including both CS and PS services) scheduled for completion in year 1999

Release 2000: 3GPP specified release of complete technical specifications for the definition and development of telecommunication services (including both CS and PS services) and also IP-based multimedia services (as defined in this TR) scheduled to be completed by the end of year 2000

Subscriber: a subscriber is an entity that has a subscription with an operator/service provider for the provisioning of specific services. The subscriber is also responsible for paying the bill for the services utilized

Supplementary service: a supplementary service modifies or supplements a basic telecommunication service (cf. 22.004 [7]

Teleservice: the services identified in 22.003 [6].

User: a user is an entity associated with a subscriber that is capable of using the subscribed services

VoIP: a voice call established over an IP based transport network and IP based control

3.2 Abbreviations

For the purposes of this TR the following abbreviations apply:

CAMEL	Customised Application for Mobile Enhanced Logic
CS	Circuit Switched
HLR	Home Location Register
IP	Internet Protocol
IPT	IP Telephony
MExE	Mobile Station Execution Environment
OSS	Operator specific services
PS	Packet Switched
QoS	Quality of Service
SAT	SIM Application Toolkit
VHE	Virtual Home Environment
WAP	Wireless Application Protocol
WTA	WAP Telephony Application

4 High level vision

4.1 The IP vision

The communication industry is going through a period of explosive change, which is both enabling and driving the convergence of services. Data is becoming a more significant proportion of traffic compared to voice. Organisations and service providers are seeking ways to consolidate voice and data traffic platforms and services. With a number of technological solutions to choose from, the Internet Protocol (IP) is today considered the most promising platform on which to build the new integrated services.

The ease of developing new applications together with IP's ability to communicate between different networks has led to IP being seen as a convergence layer that promises to evolve from a mere data platform to a provider of a much larger variety of services. An increasing demand for bandwidth, connectivity features and economy that can not be supplied by the CS mobile networks in the present form, is leading the mobile telecommunications world to reinvent itself via IP.

The IP protocol has opened up a whole range of communication applications, which may allow operators to develop totally new value added services as well as to enhance their existing solutions. The open architecture and platforms supported by the IP protocols and operating systems will lead to applications and new opportunities that are more difficult to replicate using a standard switched centralized solution. Thus, the main drivers for IP services are new services as the plain voice telephony is gradually moving to multimedia. IPT is seen as very important step forward to the mobile information society.

All IP offers the operators a complete solution for IP Telephony (IPT) and multimedia. The solution, based on the 3GPP Release 2000 standards, consists of terminals, GERAN or UTRAN radio access networks and PS domain evolved core network.

A major part of the evolution of new applications is foreseen to be in IP multimedia based services. One of the main objectives for 3GPP specifications is therefore is to ensure that the availability and behaviour of these applications when used via the 3GPP mobile access is at least as good as when used via other mobile access types.

4.2. Network evolution to an All IP network

Operators may want to migrate towards an IP based network architecture. Since the transition to an All IP networkwill not happen overnight, both traditional mobile circuit switched telephony and IP based services need be supported by a single network simultaneously. It is believed that circuit switching will live for many years together with IP services. Also there will be a large number oflegacy terminals to be supported. Also, because of real-life limitations on how quickly change occurs in networks and the mix of terminals in the network, operators may find that they must have an architecture to support different kinds of terminals and roaming between networks. It is unlikely that all networks will develop at the same speed. Hybrid architecture may be best for the majority of the operators because it allows low-risk evolution from the current networks, while enabling a full service offering.Release 2000 shall support service offerings being independent from transport technology.

The 3GPP release 2000 shall be based on an evolved Release 1999.

4.3 User perspective of services

GSM (and UMTS) succeed in a competitive marketplace due to the consistent provision of a rich diversity of high quality services. The enabling mechanisms which allow deployment of these services is transparent to the user. This is shown in Figure 1 below.

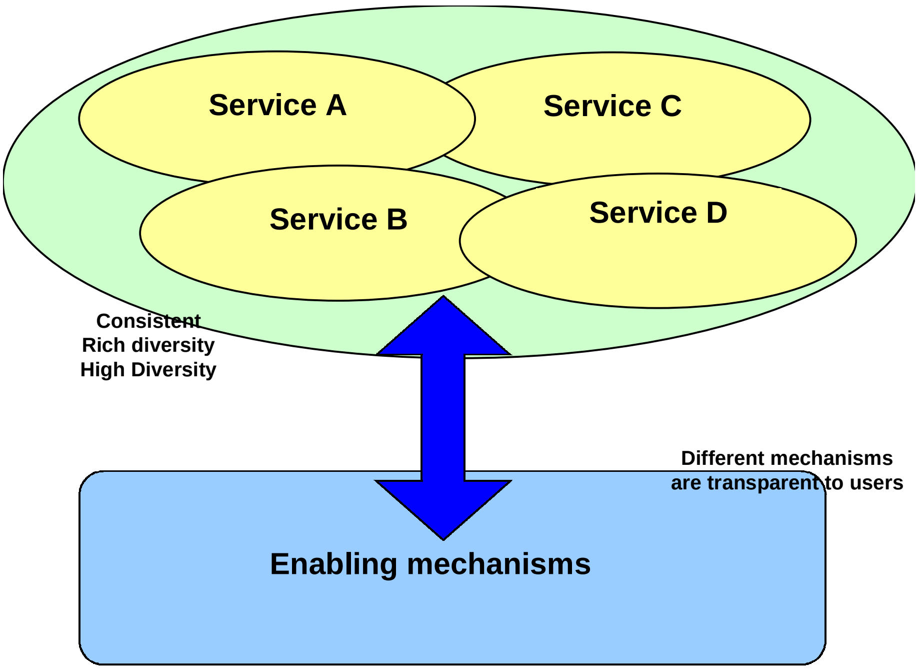

Figure 1. Transparent provision of services

Services may be categorized as basic, supplementary, operator specific, or multimedia. These categories of services may be transparent to users. Different enabling mechanisms may be used to provide services. This is shown in figure 2.

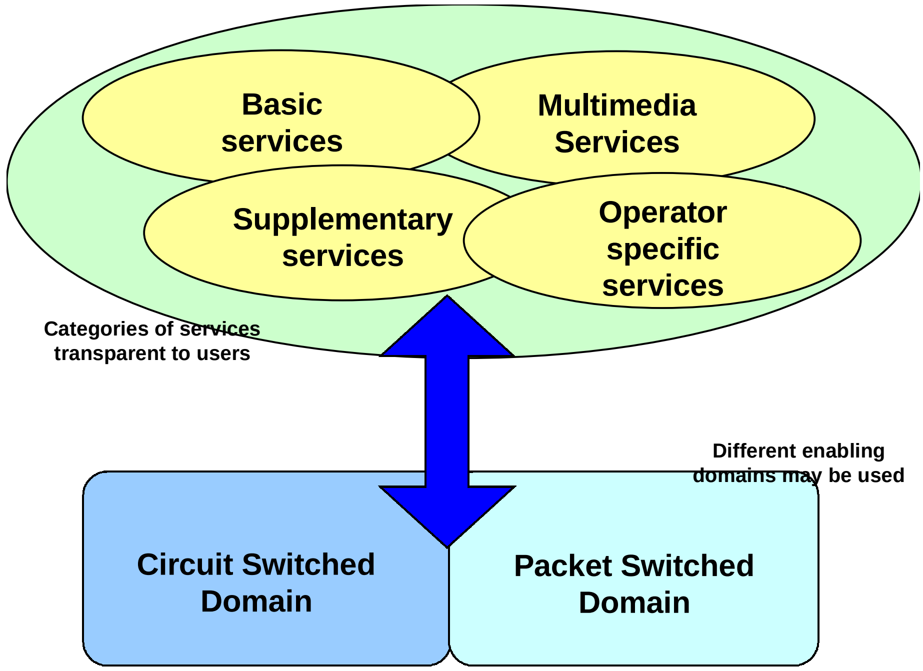

Figure 2: Different types of services and enabling domains

With succeeding releases, new and improved services and enabling mechanisms are developed and deployed. In general, most users not experience a reduction in the available service set, or degradation in the quality of the offered services. This is depicted in Figure 3.

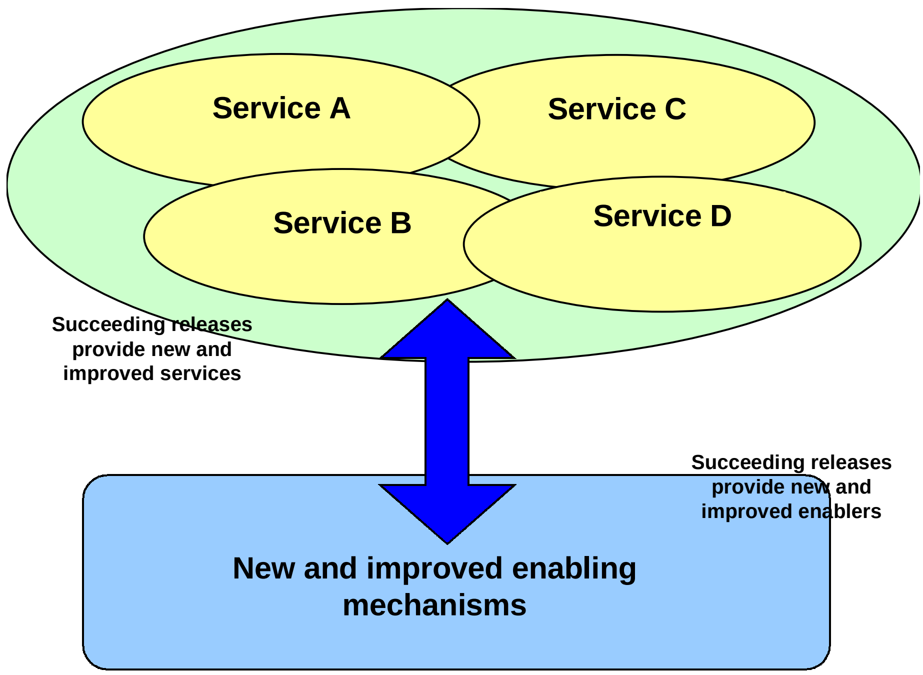

Figure 3: Succeeding releases provide new and improved services and enabling mechanisms

In Release 2000, new and improved enabling mechanisms and services will be made available. Additionally, a network option will enable the provision of services without using circuit switched enablers (the All-IP network option). In this case, the set of services available to the user, and the quality of the offered services will be no less than that available in Release 2000 networks which use circuit switched enablers. This is shown in figure 4.

Editor's note: some operators prefer identification of a minimum set of CS services in the PS domain, whereas others preferred support of all Release 99 services. Requires agreement and consensus.

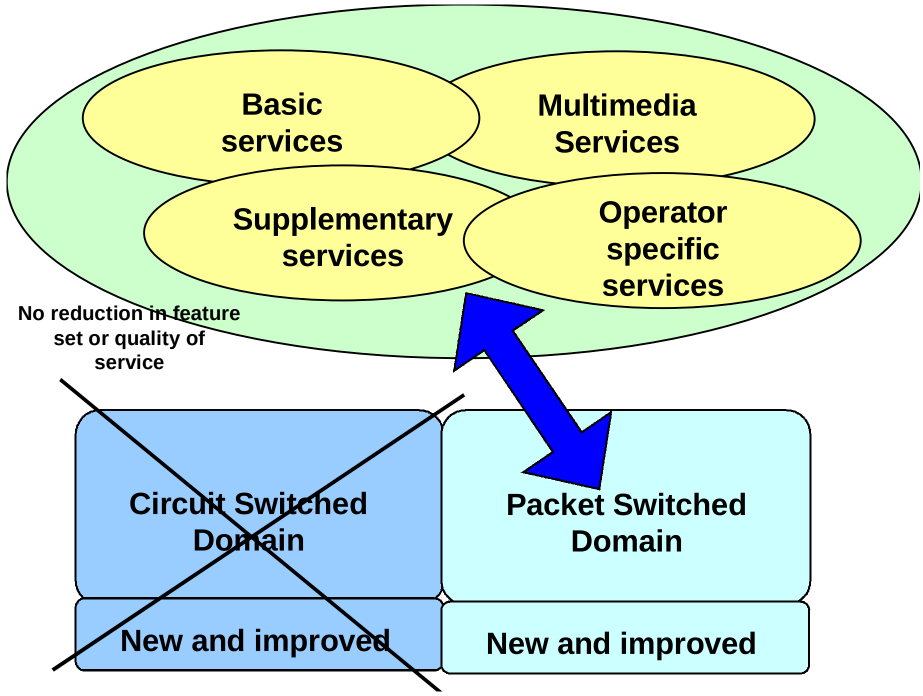

Figure 4: All-IP network option

4.4 Teleservices and multimedia services

A Release 2000 network may have both a circuit domain and packet domain, or a hybrid circuit domain and packet domain network infrastructure. In addition to teleservices available from Release 99 new services, termed multimedia services, will the available as part of Release 2000. Multimedia services may also enable enhanced usage and management of teleservices. The relationship between these tele/multimedia services, and the circuit/packet transport domains may be logically depicted as shown in Figure 5.

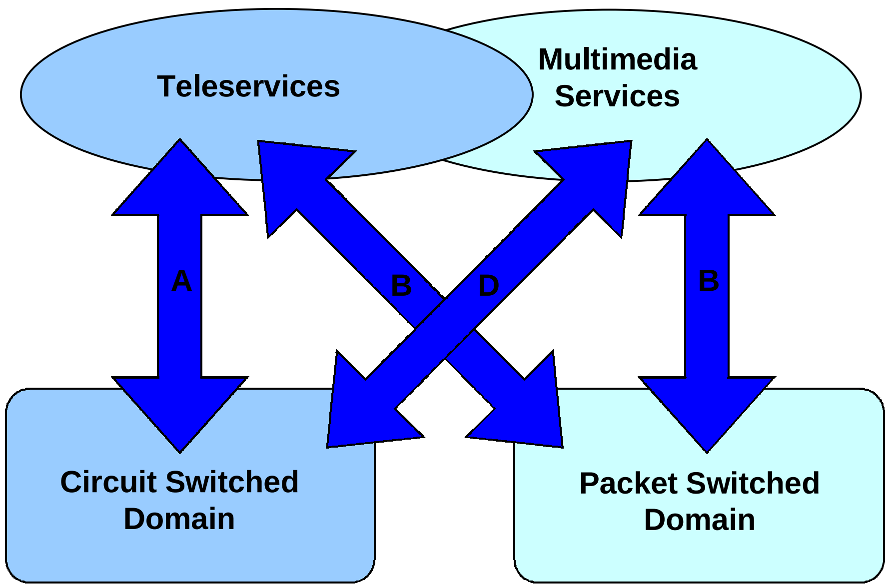

Figure 5: Teleservices and multimedia services

The logical relationship between the teleservices and multimedia services to the circuit and packet domains is subsequently described.

The “A” relationship refers to the existing relationship between the teleservices and the circuit switched based domains.

The “B” relationship refers to the support of teleservices in a packet domain. The same set of end user services may be provided across both the “A” and the “B” relationships. The existence of “B” relationship would be transparent to the end user from both a service capability and a user interface perspective. The “B” relationship could be a path for the evolution of GSM to packet based (IP) networks.

Editor's note: to verify from S2 that circuit services may be provided by the PS domain (e.g. offered by MSC servers)

The “C” relationship refers to the relationship between the multimedia services and packet domain. The “C” relationship is not merely the evolution of the 2G services and mobile terminals to the 3G environment, but also represents a new category of services, mobile terminals, services capabilities, and user expectations. Service Providers are not required to provide the existing supplementary services of the “A” and “B” relationships across the “C” relationship, although some comparable services (e.g. emergency services) may be required. Any new multimedia service which may have a similar name or functionality to a comparable standardised service, does not necessarily have to have the same look and feel from the user's perspective of the standardised service. However, the “C” relationship should provide sufficient capabilities to allow a Service Provider to develop and implement Release 2000 versions of these services that would have the same user inteface and quality of service to the end user. Voice communications is one, but not the only, real-time multimedia service that would be provided across the “C” relationship.

The “D” relationship refers to the relationship between the multimedia services and circuit domain (e.g. H.324 supported in Release 99).

4.5 PS Domain Network Requirements

The PS domain shall support both transparent and non-transparent bearer services. Account shall be taken of the need for efficiency (e.g. over the air interface, potential use of header stripping/compression techniques) in the all IP network architecture.

The PS domain shall support simultaneous realtime multimedia services and non-realtime packet services.

Editor's note: the relationship to existing solutions (e.g. in GRPS) needs to be identified

In defining PS domain requirements, alignment with multimedia developments in the wired networks should be taken into consideration, including the provision of Voice over IP, making use of the same definitions and approaches wherever possible.

In many cases, the PS domain will need to interwork with circuit switched domain, PSTN and the ISDN. These interworking requirements (including end to end quality of service issues) should be identified.

Editor's note: reference to 22.060 for existing bearer service requirements, are they sufficient?

A single call control protocol for the PS domain is recommended.

4.6 High level service requirements

Introduction of new technologies should improve the user's service experience (i.e. should not impose a reduction in the service set available or a reduction in the quality of service). New technologies should be introduced in a manner allowing for a smooth transition from existing technologies (providing a clear and smooth evolution path).

The following high level service requirements need to be incorporated for successful deployment of services based on an all-IP network. These should be considered when identifying more specific requirements.

All-IP networks shall:-

1. In general, provide backwards compatibility with the services offered by the Release 99 standard (including teleservices, supplementary services, and operator specific services)

   a) The “C” relationship in Figure 5 should provide sufficient capabilities to allow a Service Provider to develop and implement Release 99 services that would be transparent to the end user. However, Release 99 services implemented across the “C" relationship may utilise the additional capabilities of the Release 2000 multimedia environment to provide the end user with enhanced capabilities and improved user interfaces.

   b) Not all existing Release 99 services need to be supported in the PS domain (see Annex A). Some exceptions may exist, as identified in the provided feature list. A minimum set of PS domain service capabilities should be defined to enable roaming.

   Editor's note: DeWayne Sennet (AWS) to provide rephrasing of above text

   c) To enable service compatibility and access independence, it shall be possible to implement mainstream IP based multimedia (supplementary) services to be compatible with the same services when used via other types of accesses, e.g. via fixed lines (see Annex A).

2. Enable provision of services with the same (or greater) quality of service as circuit switched services.

   a) It shall be possible to offer services over an All-IP network with a quality of service that is no less than that already experienced by users of existing circuit switched networks.

   b) The enabling mechanisms (transport technology, etc.) should be transparent to the user.

   c) The All IP network shall have the ability to provide, on an end to end basis, when interworking with other All IP networks, other access networks (e.g. non-All IP), PLMNs or PSTNs, a Quality of Voice at least as good as that achieved by the Release 2000 circuit-switched (e.g. AMR codec based) wireless systems.

   Editor's note: need to rework phrasing of above…

   Editor's note: will need to separate these requirements in the feature list…

3. Shall provide the same (or greater) degree of privacy, security, and authentication as Release 2000 circuit switched services

4. Support roaming between All-IP networks and non-All-IP networks. The specific roaming scenarios required are identified in the feature list.

   Editor's note: will need to elaborate roaming, handover and cell reselection…

   Editor's note: will need to elaborate this in the feature list…

5. Support the possibility to offer a set of Release 99 services to Release 99 terminals

   Editor's note: text to be provided by Tomas Ahnberg (Telia)

5 Applicability of existing toolkits

This clause reviews the applicability of the existing toolkits in Release 99.

5.1 CAMEL

Release 2000 shall incorporate CAMEL improvements following Release 99 (e.g. Phase 4).

Users shall be able to use their existing CAMEL services in a consistent manner in Release 2000 networks. This should occur in a transparent fashion and the user need not be aware of whether the service is either circuit switched or packet switched. The same look and feel of the service should be maintained.

Users should be able to indicate their service preferences (e.g. ring tone for specific callers) only once and the service should again be provided irrespective of network domain.

Operators shall be able to re-use their existing CAMEL services in the All IP network (cf. 21.978 [13]).

The development of new CAMEL services shall be supported independently of the network domain. Thus applications developed on CAMEL platforms shall be provisioned to users and be supported in both the packet switched and circuit switched domains in a seamless fashion.

Editor's note: CAMEL/multimedia interaction needs to be considered…

5.2 MExE

Release 2000 shall incorporate improvements made in MExE Release 2000 (see 22.057 [9]), building on the (U)SIM certificate support, security and QoS management advances made in MExE Release 99. MExE supports both WAP and Java classmark devices.

MExE Release 99 provides the ability for operators, handset manufacturers and third parties to download applications, service logic and content into MExE terminals from servers. These entities will require that it shall be possible for applications, service logic and content downloaded in Release 99, shall also be downloadable and executable in a consistent manner in a Release 2000 environment. Further, it shall be possible to do so, without the need to redevelop the MExE services in order for them to be supported in the packet domain.

MExE terminals interact with the servers using capability negotiation, and it shall be possible to continue usage of the capability negotiation in the packet domain.

Editor's note: MExE/multimedia interaction needs to be considered…

5.3 SAT

Editor's note: input awaited…

5.4 VHE/OSA/???????????????

Editor's note: input awaited from Tomas Ahnberg (Telia)

6 New service capabilities and end user benefits

By analysing and categorising existing circuit switched domain services, this clause concludes that these services shall not be re-standardised in the PS domain. This results in a basic set of services which must be available in the PS domain, with the remaining services offered in the PS domain in non-standardised ways using the service capabilities.

This clause:-

1. groups the main circuit switched domain supplementary services into several main categories of services
2. analyses each main category of services
3. identifies which services can be provided in a non-standardised way by the CAMEL, MExE and SAT toolkits
4. identifies which services (such as authentication, CAMEL call triggers etc.) which will require to be standardised

In conclusion only a minimum set of services requires to be standardised in the PS domain, with the Virtual Home Environment's CAMEL, MExE and SAT toolkits using basic primitives to create alternative, personalised call handling services tailored to the user requirements.

Editor's note: further contributions/comments/analysis welcomed…

6.1 Main categories of services

An analysis of the broad classes of existing services results in the following list of example categories:

- Basic Call

- Call Barring (includes advice of charge, prepaid)
- Call Diversion
- Call Manipulation (Call hold, transfer etc)
- Addressing (includes number translation, number portability, emergency call)

This is not a complete list of supplementary services, which is elaborated in the feature list. The above categories are further elaborated below and are then mapped into service classes.

6.2 Basic Call

This category of services provides the ability to make voice, emergency and data calls to other terminals, even when roaming. This includes interworking with existing voice and data networks for both fixed and mobile users, addressed using the standard E.164 phone numbers. This category of services must also include capabilities for Mobile Number Portability and Lawful Interception (voice and data). Implicitly, Tandem Free Operation is also included. This list is not exhaustive.

Editor's note: further clarification required, and application to the roaming case need to be clarified.

6.3 Call Barring

This category of services performs two basic functions:-

- restricting the user to subscribed services (e.g. no roaming, no long distance, session barring, bearer barring, QoS etc.)
- subscriber cost control

The requirements for call barring are oriented less at limiting those teleservices which a subscribed is subscribed to, and more at simple filtering of defining which services are made available to users (by defining their menu options on the terminal) and blocking of specific teleservices to number ranges.

Specifying which services are available to users can be done through WAP, MExE, SAT and CAMEL toolkits. Outbound call barring for teleservices can be implemented using SIM Toolkit based on the number dialled, serving network (e.g. CAMEL), and MExE applications (e.g. MExE services, WAP WTA applications etc.). Therefore, by using service capabilities there is no requirement for a specific call barring service to be standardised in Release 2000.

6.4 Advice of Charge

A related feature in 2G networks is advice of charge, which is based on the serving network being aware of the teleservice in use, the price for it and the mark-up used in the home network.

In future, the price charged to a subscriber may bear little relation to the charges imposed by the serving network because there may be special offers/discounts, or the service may include elements charged elsewhere (e.g., content charged by a 3rd party).

Therefore the advice of charge may originate from the (mostly IN-based) pre-pay system and delivered to the user by various means. A method of displaying the received charging information to the user could be WAP/MExE.

6.5 Call Diversion

This category of services include immediate call diversion, call diversion on no reply, call diversion on not reachable, call diversion on busy and call completion to busy subscriber.

With terminals becoming more sophisticated, they should be capable of receiving alerts about incoming calls even when already engaged in a call, allowing terminals to signal how to process with the call. Therefore the requirement for call diversion features within the serving network is substantially reduced.

The features which are required in the serving network (e..g. provided by CAMEL/HLR) are:

1. immediate call diversion
2. call diversion on no reply
3. a set of primitives that allows the terminal to:
    a) be notified of incoming calls (including when already engaged in a call)
    b) hold/transfer/accept any of the incoming calls to another destination
    c) be notified of success or failure of these actions

Therefore no basic call divert features other than those proposed above are required to be standardised (through HLR/CAMEL).

This allows terminals to be capable of providing the call diversion features when reachable (with tbe service logic securely downloaded using MExE, building on WAP's WTA where available), and the home network/serving network to handle call diversion when the terminal is unreachable (using HLR/CAMEL). The basic primitives can also provide capability to offer call waiting, hold and transfer features. Which should be accessible through the MExE toolkit.

6.6 Conferencing

A solution for basic 3-way voice conferencing supported in the network is likely to be required, since this is the most likely service to be used with the greatest gains. Although, H.323 terminals (PCs) currently offer multimedia conferencing on the terminal, it requires one terminal to "anchor" the session and provide the mixing of sessions. A more sophisticated multimedia conferencing solution may be required, but this will be dependent on the codecs used etc. It may be more appropriate to provide this as a native IP service internal or external to the network.

Multicast and broadcast support may also be offered, using IP multicast. The benefits of this approach are most likely to occur where many users are receiving the same feed on the same cell, and the commercial benefits for this are yet to be fully understood. Any IP service should allow multicast connections to be made outside of the cellular part of the core network.

6.7 Number Portability

This category of services is one of many aspects of addressing and routing which must interwork with the existing fixed and mobile 2G network schemes already deployed. Essentially number portability can be implemented within the HLR, as part of the initial inbound call-processing query. More sophisticated schemes, which resolve one or more identities to actual routing codes may overlay these 2G schemes. Parallels may be seen in the e-mail world, where e-mail addresses appear to be portable between computers with different IP addresses, and where multiple e-mail addresses per user and per device are supported.

6.8 Service Provisioning

The range of new services created in 3G will require provisioning and configuration by users and service providers. Since the range of services and the services themselves are not standardised in 3G, the specific feature codes to provision, enable and configure them cannot be standardised either. Instead, it is expected that service capabilities, personalised Internet web pages or direct access to customer helpdesk by voice telephone will be used to allow (self)provisioning, configuration and enabling of VHE services.

6.9 Summary of required service primitives

In order to build a set of supplementary services (suitable for service provider differentiation) in the VHE, a basic set of service primitives is required which are available for access from the terminal, together with call handling for those cases where the terminal is offline or unreachable. These are:

1. Authentication (as per GSM/GPRS via SIM Card)
2. Basic Call (including implicit Mobile Number Portability, Lawful Intercept and Tandem Free Operation). Shall allow interworking as voice only call with legacy networks.

   Editor's Note: need to restructure document to identify all the required services in one common subclause…
3. Internet Access (i.e., standard GPRS service)
4. Call Diversion Immediate Call Diversion on unreachable (i.e. when terminal does not respond to paging)
5. Call manipulation primitives from the terminal:
   a) set-up outbound basic call
   b) notify/accept/answer incoming call
   c) hold
   d) transfer
   e) divert
6. Call triggers (CAMEL) to monitor and manipulate multimedia calls from the home network:
   - incoming call arrival (similar to CAMEL in circuit switched domain)

The PS domain shall use the following minimum service capabilities to build and support all other services:-

- HLR/CAMEL support in the serving network
- SIM Toolkit (minimum terminal requirement, Class FFS)
- MExE Classmark 1 (minimum terminal requirement)

7 Service continuity and new services

7.1 Service continuation

Service continuation from an end user perspective is understood to be an important driver for established users of 2nd generation mobile communications systems to stay with their existing operator while moving into the 3rd generation. It is therefore important to enable operators to offer such service continuation into Release 2000. Existing Release 99 services must as a principle be supported also in Release 2000, and that any exception explicitly identified.

Note: To enable service continuation, the existing Release 99 supplementary services (refer to [7]) shall be supported also in Release 2000 for circuit services.

7.2 No new standardised supplementary services

Operators and service providers are expected to offer new (supplementary) services to their users. If those new services would be standardised then they would have to be implemented also in all visited networks, as the existing way of implementing these services have been by downloading profiles and/or CAMEL triggers to the visited network. This creates dependencies to the visited networks which makes it more difficult to deploy new services, especially operator specific ones. We have seen in GSM that this can take time which would lead to inflexibility and delays in service offerings.

To avoid such limitations the Release 99 principle of not adding more 3GPP standardised supplementary services shall be kept. Operator specific services shall instead be implemented using VHE service capabilities, as stated for Release 99.

New supplementary services shall, as a principle, not be standardised, but instead be implemented using VHE.

7.3 Service compatibility with mainstream IP based services

It is important that Release 2000 supports evolving main stream IP based multimedia services and applications. The requirement for access independence implies that such multimedia services, e.g. an IP web based Call Forwarding service where a user could access a web page to manipulate her Call Forwarding settings based on any number of input parameters, also have to be compatible with the same IP web based service supported via other accesses such as, e.g., fixed lines. The end user shall thus experience the same service behaviour irrespective if the access is made via 3G networks or a fixed line.

Many similar (supplementary) services applicable for the evolving IP based multimedia services are as a principle different from the existing GSM standardised supplementary services, see the example of the web based Call Forwarding service above.

If a choice has to be made for the Release 2000 IP based multimedia services, between being compatible either with main stream IP based services or with existing GSM standardised supplementary services, the service compatibility with main stream IP based services must be a first priority.

Editor's note: should it be an operator or customer choice to choose "classic" service compatibility? The above paragraph may require to be revisited

7.4 Support of Release 99 supplementary services in PS Release 2000

Having established the requirement for Release 99 supplementary services in Release 2000, it is required to clarify how these services are supported in.Release 2000. Release 2000 shall not specify in detail how Release 99 services are implemented in Release 2000, but solely identify the requirement for their support.

This requirement allows the Release 99 services requirements to be fulfilled using the VHE service capabilities of Release 2000.

8 Case study of realisation of some services

e.g. CFU

Editor's note: contributions invited

9 Evaluation of what does and does not need to be standardised by 3GPP

To promote this access independence for IP based services it is necessary for Release 2000 to support and follow main stream IP-based multimedia standards, such as H.323 and SIP. This also means that 3GPP shall not standardise any mobile specific extensions to these standards.

There are cases where today’s IP based standards have to be modified to suit the mobile environment of R2000. Such modifications shall then be done by enhancing those IP based standards themselves in their relevant standardisation fora. As users can be expected to require access independence to have their services available anywhere and anytime, and as mobile communications are becoming more and more important, such mobile specific modifications in main stream standards should be achievable.

To promote access independence for IP based services Release 2000 shall support and follow main stream IP-based multimedia standards, such as H.323 and SIP. No mobile specific extensions to such standards shall be standardised by 3GPP. Where today's IP based standards have to be modified to suit the mobile environment of 3G, such modifications shall instead be done by enhancing those IP based standards themselves.

10 Interoperability requirements

Editor's note: contribution expected from Telia

11 Release workplan

In order to clearly state the TSG-S1 Service Requirements to other TSG's and WG's in a timely fashion the following Work Plan is proposed.

S1	***Dates***	***Actions***
S1#7	Feb 9-11,2000	• Work on TR22.976 so it is suitable for v1.0.0 at SA#7 • Liase TR22.976 to S2#12.
S1#8	April 10-14, 2000	• Prepare TR22.976 for approval at SA#8 • Liase TR22.976 to S2#13. • Work on of any new Stage 1's required so they are suitable for v1.0.0 at SA#8. • Produce initial CR's to the existing 22-series
S1#9	July 17-21, 2000	• Prepare any new Stage 1's for approval at SA#9. • Complete CR's to the 22-series.
S1#10	November 13-17, 2000	• Revise Stage 1's in line with feedback from other TSG's and WG's. • Begin TR on R2001.

Editor's note: need to also consider workplan for subsequent releases

Annex A PS Domain feature list evaluation for release 2000 (Normative)

Key to Table

E = Essential for release 00, launch of a commercial all-IP network is not viable with these missing or required in R00 terminal specifications to enable forward compatibility to future releases

D= Desirable for Release 2000, important features to enable a competitive and successful service launch, but could be slipped to Release 2001

R99 = Features already supported in Release 99, shall also be supported as part of Release 2000

R01+ = could wait for these features, but hooks are required in Release 2000 to enable them to be added later

No = Not needed in the PS domain

Note: Circuit switched domain services are not considered at this annex. S1 has agreed to maintain the existing Release 99 requirements in Release 2000, allowing full service continuity.

Note: All Release 99 features shifted by any reason to Release 2000 shall be included as E R’00 (to be verified feature by feature)

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
DTAP CC							i.e. 04.08 based CC. Seamless support for existing GSM services This set of requirements means 04.08/04.80 CC and SS in PS domain
Rel 99 CS terminal and circuit service support by circuit switched domain.	Support of R99 TS11, TS12, BS20, call offering SS, call completion SS, call restriction SS, CCBS SS, number identification SS etc.	X					How this requirement is supported (e.g. IP Transport, MSC servers, etc) in the circuit switched domain is out of S1 scope.
Rel 99 CS terminal and circuit service support by PS Domain.	Support of R99 TS11, TS12, BS20, call offering SS, call completion SS, call restriction SS, CCBS SS, number identification SS etc.						This requirement means 04.08/04.80 CC and SS support by PS domain.

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
IP CC							e.g. H.323 / SIP related IP multimedia services.
IP multimedia services							
IP telephony	Single medium IP voice call (using H.323 or SIP) with end user perceived quality equal or better than 2G GSM voice call						MMI must be identical to the standard telephony MMI (dialled digit, off hook, connection, on hook). Including end to end QoS support.
Multimedia IP Call	Includes IP telephony, all real time calls single and multi-media, processed by IP CC						Including end to end QoS support. It is desirable to limit the standard to one protocol only. S2 should decide on the standard protocol. The usage of any additional protocol may be based on the network transport function (bearer service).
Emergency Voice Call	Basic emergency voice call over IP						This must use the existing emergency numbering schemes (22.101). Must be compliant with FCC mandates, European and other regulatory requirements.
Group calling	This requirement covers various group call services (e.g. PMR/ASCI type of services)						Service requirements FFS
Short message service (CBS)	As specified by 23.041.	X					No additional standardisation work required for S1
Short message service (SMS PTP)	As specified by 23.040.	X					No additional standardisation work required for S1
Multimedia messages (MMS)	Support of multimedia messaging in PS domain	?					Currently supported in R99 for MS/MS. Need to consider messaging to and from other access.

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Facsimile service							
Store and forward	Transfer of text or images from a MS to a store and forward unit for subsequent delivery to a fax machine. Faxes from PSTN/ISDN to mobile terminals are stored in a store-and-forward unit.						Support of bearer service is, however, necessary to allow customised solutions to be implemented (based on T.37 and/or T.38).
End-to-end	End-to-end fax between a PSTN/ISDN fax machine and a mobile terminal.						
Services independence from transport technology	Possibility to operate in different transport environments (e.g. all IP or other different from IP)						
IP bearer services							
Point-to-Point	As in 22.060	X					No additional standardisation work required for S1
Point-to-Multipoint	PTM services such as PTM-Multicast, PTM Group Call, IP Multicast, IP Distribution Services (MDS)						Implementation of PTM services for example as defined within GPRS specifications
Asymmetric bearers	Separate parameters at the User Interface for the uplink and downlink data rate and QoS. This feature is already within R99 (e.g. TS 23.107)	X					No additional standardisation work required for S1

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Support of QoS mechanisms for real time services		?					QoS is also part of R99, but the features for real time conversational services might be delayed
Multicall capability	Support of multiple active PS sessions (TS 22.060)	X					No additional standardisation work required for S1
Interworking							Including end to end QoS support
IPv4 interworking		X					Same reqs as Rel 99 GPRS
IPv6 interworking		X					Same reqs as Rel 99 GPRS(?)
Speech to/from PSTN / ISDN / 2G CS mobile / 3G CS mobile	Full interoperability between corresponding services in PSTN/GSM environment and UMTS rel00 environment						
Multimedia to/from Internet-H.323	H.323 protocol interworking						Incl. Intranet.
Multimedia to/from Internet – SIP	SIP protocol interworking						Incl. Intranet.
Modem and ISDN interworking	Access to PSTN / ISDN dial up –data services						Removed from R99 at SA1#6
Interworking with ISDN multimedia applications	Service compatibility between real time single/multimedia N-ISDN applications and single/multimedia UMTS rel 00 applications including H.324						
Interworking with other access networks (e.g. cable)							

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Interworking with intranets (including VPNs)	VPN functionality (firewall bypass) shall be supported						May possibly be supported at the application layer.
Roaming	Editor's note: Roaming section not reviewed						
CS/PS GSM/UMTS R99 to R00 PS							(U)SIM and Multimode terminal Roaming. Roaming should be possible for both R99 and pre R99 GSM/GPRS networks. (Note: roaming to R00 CS is included but not within the scope)
R00 PS to GSM/GPRS							(U)SIM and Multimode terminal roaming. Roaming should be possible for both R99 and pre R99 GSM/GPRS networks.
ANSI-41 to R00 PS							
R00 PS to ANSI-41							
Handover	Editor's note: Handover section not reviewed						
Speech from PS Domain to CS-GSM /UMTS							
Speech from CS-GSM /UMTS to PS Domain							
Handover of parallel sessions/calls (with different QoS)							
Multimedia handover between R00 UTRAN and GERAN							To include real-time services in addition to R99 “best effort”

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Multimedia services to/from alternative access technologies (e.g. HIPERLAN/2)							Alternative access technologies to include HIPERLAN/2
Supplementary Services (PS Domain)							IP Multimedia / IP Telephony aware supplementary services to be considered only from end-user need view point. (Standardization, service capabilities and implementation FFS). S2 choice of MM CC protocol may support some MM services implicitly. Some of the following services can be provided at the application level (i.e. no standardisation required). The H.450 standards defines supplementary services for H.323, and SIP provides tools to build supplementary services.
Multimedia Call Barring	Enables mobile subscriber to have barring of certain categories of outgoing multimedia calls.						Also includes incoming multimedia calls and barring when roaming.
Network Barring	Editor's note: Description and justification to be supplied by Horst Rauch (T-Mobil)						FFS
Session Barring	Editor's note: Description and justification to be supplied by Horst Rauch (T-Mobil)						FFS
Bearer Barring	Editor's note: Description and justification to be supplied by Horst Rauch (T-Mobil)						FFS. Might be based on the QoS parameters.

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Multimedia Call Forwarding	Forwarding of multimedia call (e.g. triggered by conditions of Unconditional, Busy, No Reply, Not Reachable etc). Triggers/activation will be different and more detailed in a MM environment for the different media components.						Conditions require to be evaluated.
Multimedia Call Transfer	Enables served mobile subscriber who has a multimedia call, to connect the other parties in the multimedia call and release the served mobile subscriber's own connection.						
Multimedia Call Deflection Service	Enables the served mobile subscriber to respond to an incoming multimedia call offered by the network by requesting redirection of this multimedia call to another address or location.						
Multimedia Call Holding	Allows served mobile subscriber to interrupt communication on an existing active multimedia call and then subsequently re-establish communication						
Multimedia Call Waiting	Permits mobile subscriber to be notified of an incoming multimedia call while the mobile subscriber is engaged in other multimedia call(s). Subscriber can either accept, reject, ignore, or deflect the incoming multimedia call.						
Advice of Charge	Supply user sufficient information to allow real-time estimate of the call charge.						
Caller Identification and restriction	Similar to CLIP, CLIR, CNAP and CNAR. Editor's note: add to abbreviations						Could include additional IP related information such as IP address. Need to support European, FCC and other regulatory requirements.

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Connected Line Identification and restriction	Similar to COLP and COLR Editor's note: add to abbreviations						Could include additional IP related information such as IP address. Need to support European, FCC and other regulatory requirements.
Multimedia conferencing	Similar to Multiparty but is applicable to multimedia calls.						
Multimedia call-back when free / CCBS	Editor's note: definition required in the MM case						
Closed user group (CUG) / community of interest	Editor's note: definition required in the MM case						
Precedence and Pre-emption service	Editor's note: definition required in the MM case						
Network Services							
Operator Determined Barring (ODB)	Allows service providers to regulate subscriber access to services by the barring of certain categories of outgoing or incoming multimedia calls and packet services. ODB could terminate ongoing multimedia calls and could bar future multimedia calls and packet services.	?					
CAMEL Support for Multimedia Services	Provides mechanisms to support multimedia services consistently & independently of the serving network.						CAMEL enhancements, but additionally needs to support multimedia calls. Implementation is FFS. Editor's note: DeWayne Sennet (AWS) to WIN support proposal

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
(U)SIM Toolkit	Feature provides a set of facilities which allow the (U)SIM to interact with external entities (e.g. the network, the Mobile Equipment, or the user) to enable value-added multimedia applications to exist in the (U)SIM.						
OSA for new elements	Provision of an API for controlled, secure and accountable access to multimedia service capability features by applications, based on the user profile						E.g. H.323/SIP CSCF.
LCS for GPRS/PS domain	Support of LCS on the PS domain required to meet regulatory and commercial requirements (e.g., 3GPP 22.071)						Exact work required is for FFS. Need to support European, FCC and other regulatory requirements.
SoLSA	SoLSA shall facilitate user-dependent radio resource selection based on LSA (e.g. when user is located at his office, radio coverage provided with indoor radio solutions should be preferred).						
Lawful Surveillance / Intercept							Need to support European, FCC and other regulatory requirements.
Number Portability	Ability for subscriber to change service providers while retaining the original directory number. Includes mobile to mobile, mobile to landline, & landline to mobile number portability scenarios.						Need to support European, FCC and other regulatory requirements.
Mobile Station Application Execution Environment (MExE)	Provides standardized execution environment in an MS, and an ability to negotiate its supported capabilities with a MExE service provider, allowing applications to be developed independently of any MS platform.						

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Personalization of Mobile Equipment (ME)	Storage of information in the ME which limits the SIMs which will operate with the ME.	X					No standardisation work required for R99
Advanced Addressing	Support of symbolic and advanced addressing						Addressing depends on applications, e.g. e-mail addresses are used for e-mail, E.164 is used for telephony, ICQ uses IP addresses. This does not require standardisation by 3GPP.
System Selection	Ability for the mobile equipment to choose a preferred service provider, based upon geographic location, frequency band preferences, available operators, etc. Also the ability to force a mobile station to "disallow" service from a "forbidden" service provider, and to force a mobile station to use "home" services. Service provider lists must be downloadable over-the-air. The possibility for operators with multimode networks, e.g. with GSM and UMTS radio access networks, to control which RAN a user accesses.						Similar to ANSI-136 Intelligent Roaming.
Over-the-Air Service Provisioning	Ability to download parameters to either the SIM or ME for provisioning of services. This includes both subscription parameters as well as operator-specific parameters.						Similar to ANSI-136 OTASP and OTAPA.
Charging							
Implementation of on-line charging mechanisms for the support of Pre-paid services	Definition of charging mechanisms for the support of IP multimedia pre-paid services						
Event/transaction based charging mechanisms (e.g. content based)	Definition of charging mechanisms for the provisioning of IP based Value Added Services						

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Charging aspects – need to charge for each PDP context (PS sessions) independently							

.

History

Document history		
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Puerto Vallarta, Mexico, 24th - 28th January 2000

Agenda Item: 6

Source: Telia (editor)

Title: draft TR 23.821 v0.1.0 “Architecture Principles for Release 2000”

Document for: Approval

Introduction

A draft version 0.1.0 of TR 23.821 is provided for approval. The following modifications have been performed:

1. Addition of a reference architecture description in Chapter 5 (tdoc S2-000249)
2. Addition of new section 5.1 Architecture Principles, including text for 2 items (tdocs S2-99D22, S2-00252)
3. Addition of an end-to-end QoS functional model and related text (tdoc S2-99F20)
4. Added text in Ch 19 Work Plan
5. For editorial purposes, the most significant beeing:

- In the beginning of Chapters 5 & 9, statements which specifies in which R00 TS/TR document specific text is intended to be included
- Addition of “header” for text from S2-000252
- Creation of a new chapter 9.1 which now contains the text from tdoc S2-99F20

Proposal

It is proposed to approve the provided draft TR 23.821 v0.1.0.

TR 23.821 V0.~~0~~1.0 (~~1999-10~~2000-01)

Technical Report

3rd Generation Partnership Project (3GPP);
Technical Specification Group Services and System Aspects;
Architecture Principles for Release 2000
(3G TR 23.821 version 0.1.0)

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Contents

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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 TR, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version 3.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

The scope of this Technical Report is to list architectural requirements, features, functions and solutions of UMTS inside the scope of UMTS Release 00. These are working assumptions agreed by TSG SA WG2. The TR focuses on

- new/modified functionality as compared to Release 99
- technical description of the features, functions and solutions of R00.

It is expected that this TR will act as a basis for the detailed Stage 2 specification work.

This TR has been created to ease the development of R00 work prior to the finalization of the R99 specifications. In conjunction with when R99 is finalized, work on the TR will cease and the relevant CRs will be produced to incorporate the contents of this TR within the R00 version of the specifications."

2 References

[Editor's n~~N~~ote: Chapter to be completed]

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]

[2]

3 Definitions, symbols and abbreviations

[Editor's ~~Note~~note: To be completed]

3.1 Definitions

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

3.2 Symbols

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

3.3 Abbreviations

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

4 Introduction

5 Reference Architecture

[Editor's note: this chapter discusses overall reference architecture issues which are not covered in other chapters]

[Note: The following sections are intended to be included in TS 23.002 (proposed chapter, if any):

- Section 5.2 (Chapter 5)
- Section 5.3 (Chapter 4)
- Section 5.4 (Chapter 6. Alternatively, a chapter on Description of Reference Points may be created.)]

5.1 Architecture Principles

Transport Independence (to control heterogeneous bearer mechanisms): The GSM/UMTS CN reference architecture shall be independent of the underlying transport mechanism (e.g. STM, ATM or IP) further more the operators shall have the freedom to utilise a single or any combination of transport technologies.

Standardised alternatives for transport mechanisms: The alternatives for the signalling transport (e.g. SS7, SIGTRAN) for the service control, call control and bearer control protocols as well as the alternatives for the user plane transport shall be standardised for the relevant transport mechanisms.

5.2 Reference Architecture Overview

The full view of Release 2000 architecture is provided in Figure 5-1.

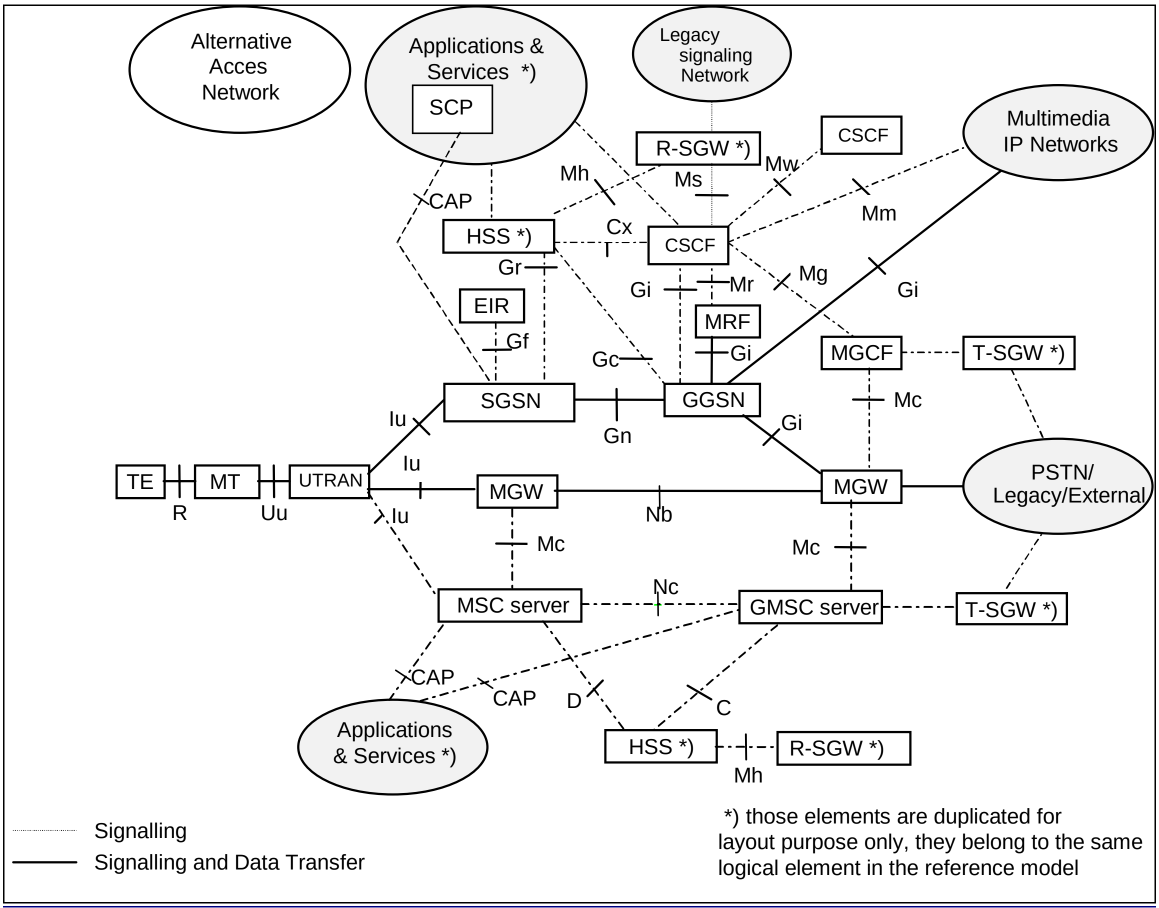

Figure 5-1: Reference Architecture for Release 2000

Note: A (G)MSC Server and associated MGW can be implemented as a single node as with the (G)MSC in R99.

[Editor's note: The final approval of Figure 5-1 and related text is dependent on e.g.:

- *The specification of R00 requirements.*
- *The relationship between different call control models (H.323/H.324 etc.) needs to be clarified.*
- *Clarification of which interfaces/reference points that require standardization and which standardized protocols (from which standard body) to use, including those that 3GPP still has work on.*
- *Addition of potentially missing reference points, e.g. Gs and a reference point between the MGW and the multimedia related control nodes (e.g., CSCF, etc.) when multimedia is going to be operated in the CS Services domain. Also, further reference points should be considered, e.g. internal to the proposed HSS as well as components of the HSS to the other nodes.*
- *Clarification of the terminology for domains (ps, cs domains vs GPRS, Multimedia, Teleservices domains).*
- *This reference architecture is subject to verification through the inclusion of flow charts showing signalling flows for MM, SM etc. in, e.g., an annex or other chapters in the TR.*
- *The inclusion of separate CBC node in the figure.*
- *Clarify the relationship with Mobile IP.*
- *GERAN has been removed from the reference architecture in Figure 5.1 since the issue of GERAN and related interfaces, i.e., Iu_ps, Gb, A are currently being handled in SMG2 and SMG12. Appropriate description on the GERAN will be added when results are available.]*

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.

5.3 Functional Elements

5.3.1 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.

Every CSCF acting as a Serving CSCF 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.

5.3.2 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).

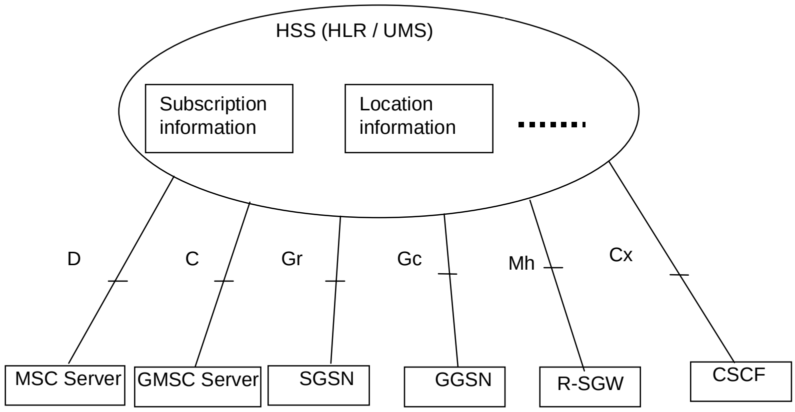

Figure 5-2: Example of a Generic HSS structure and basic interfaces

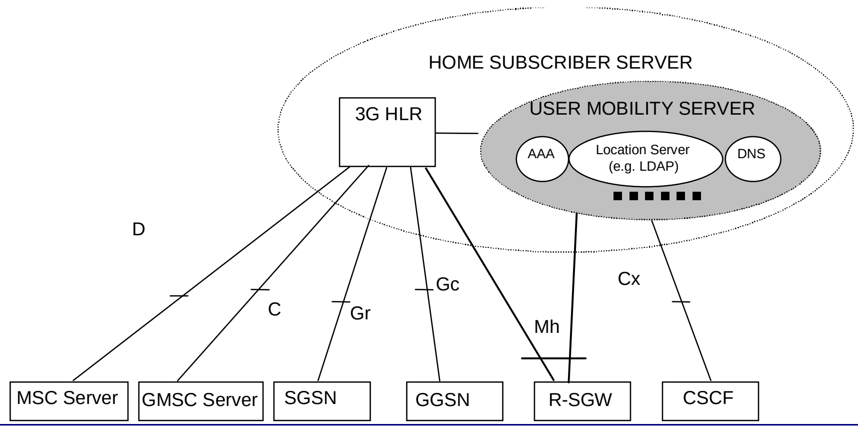

Figure 5-3: Example of HSS structure with UMS Specific Functionality

The HSS may consist of the following elements as shown in the Figure 5-3:

1) User Mobility Server (UMS): it stores Service Profile for the Multimedia domain 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.

2) 3G HLR: A UMTS HLR enhanced to support Release 2000 access specific information.

5.3.3 Transport Signalling Gateway Function (T-SGW)

This is component in the R00 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.

5.3.4 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 UMTS Teleservices domain and UMTS GPRS domain and is not involving the Multimedia 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

5.3.5 Media Gateway Control Function (MGCF)

This component 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 depending 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 network call control protocols.
- Out of band information assumed to be received in MGCF and may be forwarded to CSCF/MGW.

5.3.6 Media Gateway Function (MGW)

This component is PSTN/PLMN transport termination point for a defined network and interfaces UTRAN with the 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.

- 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.

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.

The MGW bearer control and payload processing capabilities will also need to support mobile specific functions such as SRNS relocation/handover and anchoring It is expected that current H.248 standard mechanisms can be applied to enable this.

5.3.7 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.
- Is responsible for bearer control (with GGSN and MGW) in case of multi party/multi media conference
- may communicate with CSCF for service validation for multiparty/multimedia sessions.

5.3.8 MSC Server

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

5.3.9 Gateway MSC Server

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

5.3.10 MSC

A MSC server and a MGW make up the full functionality of a MSC as defined in 23.002 Gateway MSC

5.3.11 Gateway MSC

A GMSC server and a MGW make up the full functionality of a GMSC as defined in 23.002

[*Editor's note: There is a need to consider possibilities that call incoming to the PLMN may be routed to entities other than the GMSC, e.g., for networks that do not deploy CS domain.*]

5.4 Description of Reference Points

5.4.1 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.

5.4.2 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.

5.4.3 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.

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.

5.4.4 Mh Reference Point (HSS – R-SGW)

This interface supports the exchange of mobility management and subscription data information between HSS and R99 and 2G networks. This is required to support Release 2000 network users who are roaming in R99 and 2G networks.

5.4.5 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.

5.4.6 Mr Reference Point (CSCF - MRF)

Allows the CSCF to control the resources within the MRF.

5.4.7 Ms Reference Point (CSCF – R-SGW)

This is an interface between the CSCF and R-SGW.

5.4.8 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).

5.4.9 Nc Reference Point (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 R'00 architecture different options for signalling transport on Nc shall be possible including IP.

5.4.10 Nb Reference Point (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. In the R00 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.

5.4.11 Reference Points towards SCP (CAP based interfaces)

This includes the interfaces from the SGSN to the SCP, from the MSC Server to the SCP, and the GMSC Server to the SCP.

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

The interface from the MSC Server to the SCP, and the GMSC Server to the SCP is the standard interface defined for CAMEL feature, which provides the mechanisms to support services of operators which are not covered by standardized UMTS/GSM services even when roaming outside the home PLMN.

The CAP based interfaces may be implemented using CAP over IP, or CAP over SS7 as shown in Figure 5-4.

<table>
<tr><td colspan="3" align="center">CAP</td></tr>
<tr><td colspan="3" align="center">TCAP</td></tr>
<tr><td colspan="3" align="center">SCCP</td></tr>
<tr><td align="center">M3UA</td><td align="center">MTP-3B</td><td rowspan="2" align="center">Narrow-band SS7</td></tr>
<tr><td align="center">SCTP (1)</td><td align="center">SAAL</td></tr>
<tr><td align="center">IP (2)</td><td align="center">ATM(2)</td><td align="center">STM (2)</td></tr>
</table>

Figure 5-4: Protocol Stack for CAP

Note:

(1) In IETF work is ongoing (e.g., SCTP/UDP/IP or directly SCTP/IP). The finally selected protocol stack is meant here.

(2) The protocols do not correspond to the same OSI layer. They are drawn on the same height as they are "transport alternatives".

5.4.12 Gc, Gr, C, D Reference Points MAP based interfaces)

This includes the interfaces from the GGSN to the HSS (Gc reference point), from the SGSN to the HSS (Gr reference point), from the GMSC Server to the HSS (C reference point), and the MSC Server to the HSS (D reference point).

The MAP based interfaces may be implemented using MAP transported over IP, or MAP over SS7.

MAP can be transported on the same protocol stacks as CAP (refer to protocol stack in Figure 5-4)

5.4.13 Iu Reference Point

This is the reference point between UTRAN and the R00 core network. This reference point is realized by one or more interfaces:

- Between UTRAN and SGSN, transport of user data is IP based.
- Between UTRAN and SGSN, transport of signalling is based on IP or SS#7.
- Between UTRAN and MGW, transport of user data is based on different technologies (e.g., IP, AAL2), and includes the relevant bearer control protocol in the interface.
- Between UTRAN and MSC server, transport of signalling is based on IP or SS#7.

When the Iu_cs is ATM based, then the protocols used can be based on R99 protocols or an evolved version.

When Iu_cs is IP based, new IP transport related protocols need to be added as part of the Iu protocols. It shall be possible to have R99 Iu interface with MSCs compliant to R99 specifications in the network.

It shall be possible to have a R99 CS domain with R99 Iu_cs reference point coexisting with a R00 Iu reference point.

6 Mobility Management

6.1 Location Management

6.2 Handover

6.3 Mobility across networks

[Editor's note: this section with deal with e.g. Mobile IP related issues]

7 Service Platforms

[Editor's n~~N~~ote: this chapter deals withVHE/OSA, SAT, CAMEL etc.]

7.1 Location Services

8 Multimedia

[Editor's note: this chapter deals with multimedia related issues such as H.323, H.324, their impact on the reference architecture etc.]

8.1 Signalling

8.2 Transcoder

9 QoS

[Note: The following section is intended to be included in TS 23.107 (proposed chapter, if any):

- 9.1 (Chapter 6.2)]

9.1 QoS End-to-End Functional Architecture

To provide QoS end-to-end, it is necessary to manage the QoS within each domain. An IP BS Manager is used to control the external IP bearer service. Due to the different techniques used within the IP network, this communicates to the UMTS BS manager through the Translation function.

Whenever resources not owned or controlled by the UMTS network are required to provide QoS, it is necessary to interwork with an external resource manager that controls those resources.

IP BS Manager

The IP BS Manager uses standard IP mechanisms to manage the IP bearer service. These mechanisms may be different from mechanisms used within the UMTS, and may have different parameters controlling the service. The translation/mapping function must provide the interworking between the mechanisms and parameters used within the UMTS and the external IP bearer service.

Resource Manager

Within the UMTS network, there is resource management performed by various nodes in the admission control decision. The resources considered here are under the direct control of the UMTS network.

In IP Networks, it is also necessary to perform resource management to ensure that resources required for a service are available. Where the resources for the IP Bearer Service to be managed are not owned by the UMTS network, the resource management of those resources would be performed through an external resource management function for the IP network.

In addition, where the UMTS network is also using external IP network resources as part of the UMTS bearer service (for example for the backbone bearer service), it may also be necessary to interwork with an external IP resource manager.

Figure 9-1 shows the scenario for control of an IP service using an IP BS Manager and an external Resource Manager.

[Editor's note: The end-to-end IP bearers may have several BS Managers throughout the path, specifically the modeling of IP based BS Manager in MT and/or TE needs to be studied. This issue is FFS.]

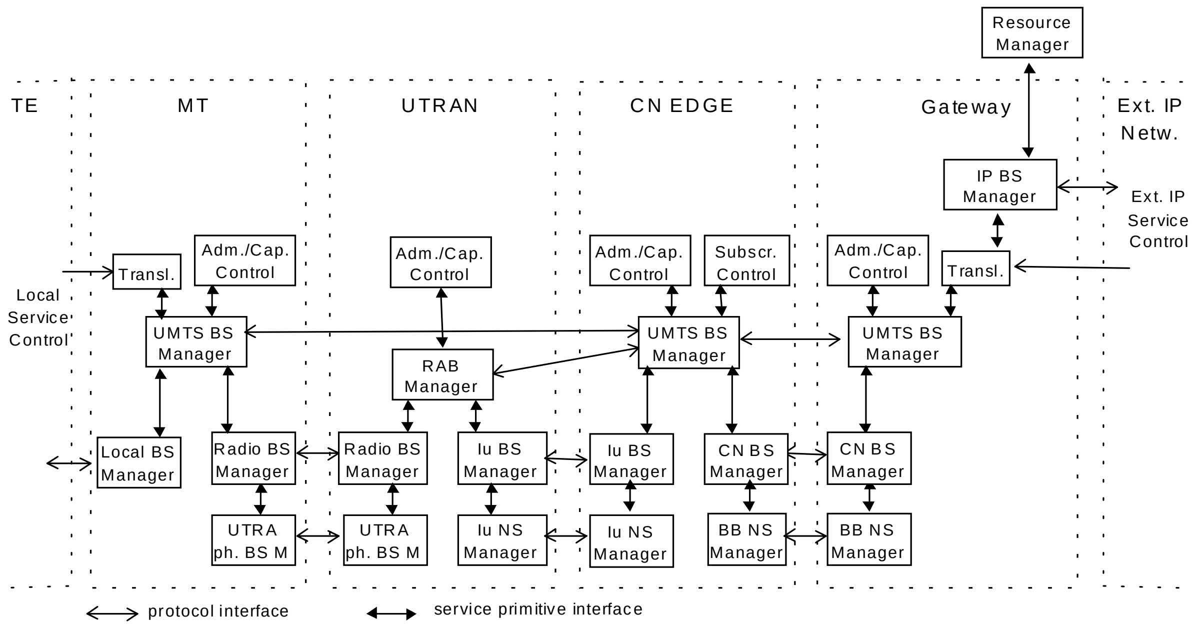

Figure 9-1: QoS management functions for UMTS bearer service in the control plane for an external IP Service

Note: This proposal does not cover the cases of a circuit switched service, or an IP service interworking with an ATM service at the gateway node.

10 Transport

[Editor's note: this chapter deals with user and control plane transport issues for relevant interfaces]

11 Point-to-Multipoint

12 Security

13 Charging

14 UTRAN Aspects

[Editor's note: requirements on UTRAN from a system perspective]

15 BSS Aspects

16 Alternative Access Networks

[Editor's note: this chapter deals with system aspects relating to other access tehnologies than UTRAN, e.g. EDGE and Hiperlan2]

17 Multi-mode

[Editor's note: this chapter deals with issues in relation to the handling of multimode terminals]

18 Compatibility

[Editor's note: this chapter deals with compabitility issues between different releases, and between different options]

19 Work Plan

[Editor's note: an overall workplan is specified here]

An overall work plan for R00 may be found in TR 30.801 v.1.1.0.

20 History

Document history		
V0.0.0	1999-10	Document created
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Network Questions:

1. What combinations of R’00 networks can exist:
   - CS Only
   - PS Only
   - IM Only
   - CS + PS
   - CS + IM
   - PS + IM
   - CS + PS + IM

2. How does the CS R’00 network differ from the CS R’99 network?

3. How does the PS R’00 network differ from the PS R’99 network?

4. Backwards Compatibility (USIMs)
   - Are R’99 USIMs supported in a R’00 terminal
   - Are R’99 USIMs supported in a R’00 network (CS+IM or PS + IM)
   - Are R’00 USIMs supported in a R’99 terminal
   - Are R’00 USIMs supported in a R’99 network (CS or PS)

5. Backwards Compatibility (terminals)
   - Are R’99 terminals (CS or PS) supported in a R’00 (CS+IM or PS+IM) network?
   - Are R’00 terminals supported in a R’99 network

6. Backwards Compatibility (services)
   - Are R’99 services (CS or PS) supported in a R’00 (CS+IM or PS + IM) network?
   - Are R’00 services supported in a R’99 network

7. Roaming
   - Can a R’00 terminal roam into a R’99 network
   - Can a R’99 terminal roam into a R’00 network

8. Handover
   - Can a R’00 terminal handover into a R’99 network
   - Can a R’99 terminal handover into a R’00 network