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Technical Report

3rd Generation Partnership Project;
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Study on Release 2000 services and capabilities
(3G TR 22.976 version 2.0.0 Release 2000)

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Contents

Foreword

This Technical Report has been produced by the 3GPP.

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

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where:

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1   presented to TSG for information;

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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 Release 2000, consisting of an evolved Release 99, and IP multimedia services.

The evolved Release 99 and the first phase of IP multimedia services of 3GPP Release 2000 are described in this TR provide the basis for the detailed Stage 1 specification work. The IP multimedia services will be developed in a phased approach and a longer term vision is presented in the report.

The focus of the TR is:

- Support and evolution of 3GPP Release 99
- High level vision of multimedia services
- Examination of potential service drivers
- New and evolved service capabilities and end user benefits
- Evaluation of what does and does not need to be standardised
- Release a Feature List of global interest and potential service candidates
- Case study of realisation of some services (e.g. CFU)

This TR has been created to ease the development of 3GPP Release 2000, 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; Circuit Teleservices supported by a Public Land Mobile Network (PLMN)"

[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]   TS 22.038: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; USIM/SIM Application Toolkit (USAT/SAT); Service description; Stage 1"

[12]   TS 22.001:" Digital cellular telecommunications system (Phase 2+); Technical Specification Group Services and System Aspects; Principles of circuit telecommunication services supported by a Public Land Mobile Network (PLMN)"

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

[14]   TR 22.129: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Service Aspects; Handover Requirements between UMTS and GSM or other Radio Systems"

[15]   TS 22.002: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Circuit Bearer Services (BS) supported by a Public Land Mobile Network (PLMN)"

[16]   TS 22.071: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Location Services (LCS); Service description, Stage 1"

[17]   TS 22.043: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Support of Localised Service Area (SoLSA); Service description, Stage 1"

[18]   TS 22.024: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Description of Charge Advice Information (CAI)"

3   Definitions and abbreviations

3.1   Definitions

For the purposes of this TR the following definitions apply:

Note: The definitions need to be cross-checked between different WGs (specifically S1/S2) for TR 21.905.

Address: an address is a string or combination of digits and symbols, which identifies the specific termination points of a connection and is used for routing.

Basic services: basic services are the teleservices and bearer services as defined in [6] and [7]

CS CN domain: comprises all core network elements for provision of CS services.

Circuit services: Telecommunication services provided to "GSM/ISDN" clients via 24.008 call control.

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

General Packet Radio Service (GPRS): IP connectivity service provided to IP clients via 24.008 session management.

Hybrid network: a telecommunications network consisting of the CS CN domain, PS CN Domain and IM CN subsystem.

IP multimedia service: IP Multimedia Services that require support on the Call Control level carried on top of GPRS (this may include an equivalent set of services to the relevant subset of CS Services).

IP telephony: a voice call that uses IP based protocols (e.g. SIP) and IP for transport of all user data and signalling data. IP telephony is an example of an IP multimedia service, which uses only a single medium.

IM CN subsystem: (IP Multimedia CN subsystem) comprises all CN elements for provision of IM services

Multimedia service: a service that handles one or more media simultaneously such as speech, audio, video and data (e.g. chat text, shared whiteboard) 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. Multimedia services may be provided using either circuit switched domain, or packet switched domain, or a combination of circuit switched and packet switched domains.

Name: A name is a combination of characters and is used to identify end users.

Number: A number is a string of decimal digits. An E.164 number (as defined in E.164) can act in the 'role' of both a name and an address. When an E.164 number is used in a mobile environment it is acting in the role of a name only.

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

PS CN domain: comprises all core network elements for provision of GPRS.

PS services: The superset of IM services and GPRS

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) including IP multimedia services (as defined in this TR) scheduled to be completed by the end of year 2000

Service continuity: handling of changes (e.g. bearers) in the network such that the user is provided with the experience of an active service (e.g. voice) being seamlessly maintained.

Serving network: the network to which the subscriber is currently registered

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

3.2 Abbreviations

For the purposes of this TR the following abbreviations apply:

3GPP	3rd Generation Partnership Project
BDN	Barred Dialling Number
AoC	Advice of Charge
CAMEL	Customised Application for Mobile Enhanced Logic
CC	Call Control
CDR	Call Detail Record
CCBS	Completion of Calls to Busy Subscriber
CFNR	Call Forwarding on No Reply
COLP	Connected Line identification Presentation
COLR	Connected Line identification Restriction
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CNAP	Calling Name Presentation
CNAR	Calling Name Restriction

CS	Circuit Switched
CUG	Closed User Group
EDGE	Enhanced Data rates for GSM Evolution
FDN	Fixed Dialling Number
GERAN	GSM/EDGE Radio Access Network
HLR	Home Location Register
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPT	IP Telephony
GPRS	General Packet Radio Service
MExE	Mobile station application Execution Environment
O&M	Operations and Maintenance
ODB	Operator Determined Barring
OSA	Open Service Architecture
OSS	Operator specific services
PS	Packet Switched
QoS	Quality of Service
SAT	SIM Application Toolkit
SoLSA	Support of Local Service Area
SM	Session Management
UE	User Equipment
UTRAN	Universal Terrestrial Radio Access Network
VHE	Virtual Home Environment
VoIP	Voice over IP
WAP	Wireless Application Protocol
WTA	WAP Telephony Application

4 High level vision

4.1 Phased approach for network and services evolution

The network support for IP multimedia services shall be developed in a phased approach, commencing with support of an evolved Release 99, through to the support of IP multimedia services.

4.2 Release 2000

Release 2000 shall support Release 99 specified in the 22-series specifications, as may be amended by this TR. The requirements of the Release 99 series of specifications, or their services, service capabilities and technical contents, are not replicated in this TR, however their technical requirements are automatically included in Release 2000 by reference. Further, any modification or evolution of the Release 99 as may be defined in this TR also form part of Release 2000.

Additionally, this TR identifies the requirements in Release 2000 for IP multimedia services.

4.3 The IP vision

The communication industry is going through a period of explosive change, which is both enabling and driving the convergence of services. 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 may lead to applications and new opportunities that are more difficult to replicate using a standard switched centralised solution. Thus, the main drivers for IP services are new services as the plain voice telephony is gradually moving to multimedia. IP Telephony is seen as very important step forward to the mobile information society.

Future networks potentially offer the operators a complete solution for multimedia including IP Telephony. 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 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.

4.4   Services evolution

Operators will want to offer multimedia services in a homogenous and efficient way. Since the transition to IP multimedia services will not happen overnight, both traditional mobile circuit switched services and IP multimedia services need be supported simultaneously. Circuit switching will live for many years together with IP multimedia services, and there will be a large number of legacy 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 following provides an overview of Release 2000:-

- Hybrid Architecture
    - network evolution path
- New Capabilities
    - IP based call control
    - Realtime (including Voice) services over IP with end-to-end QoS
    - GERAN (support for GSM radio including EDGE)
    - Services provided using toolkits (e.g. CAMEL, MExE, SAT, VHE/OSA)
        - Backwards compatibility with Release 99 services
        - No degradation in QoS, security, authentication, privacy
    - Support for inter-domain roaming and service continuity

The above list is not exhaustive, and is further elaborated in clause 5 Release 2000 services of this TR.

4.5   User perspective of services

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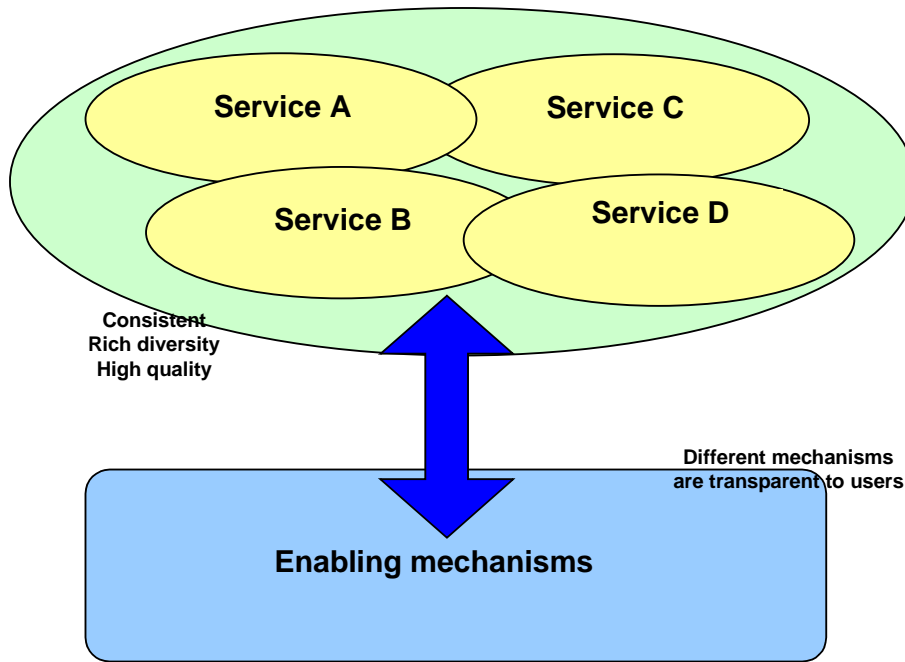
Figure 1. Transparent provision of services

Services may be categorised 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.
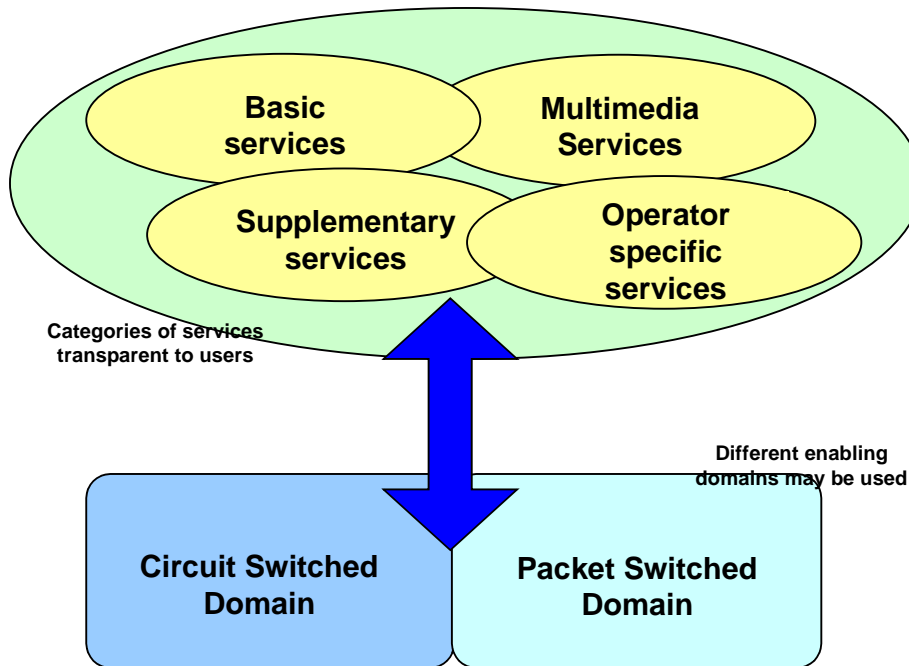
Figure 2: Different types of services and enabling domains

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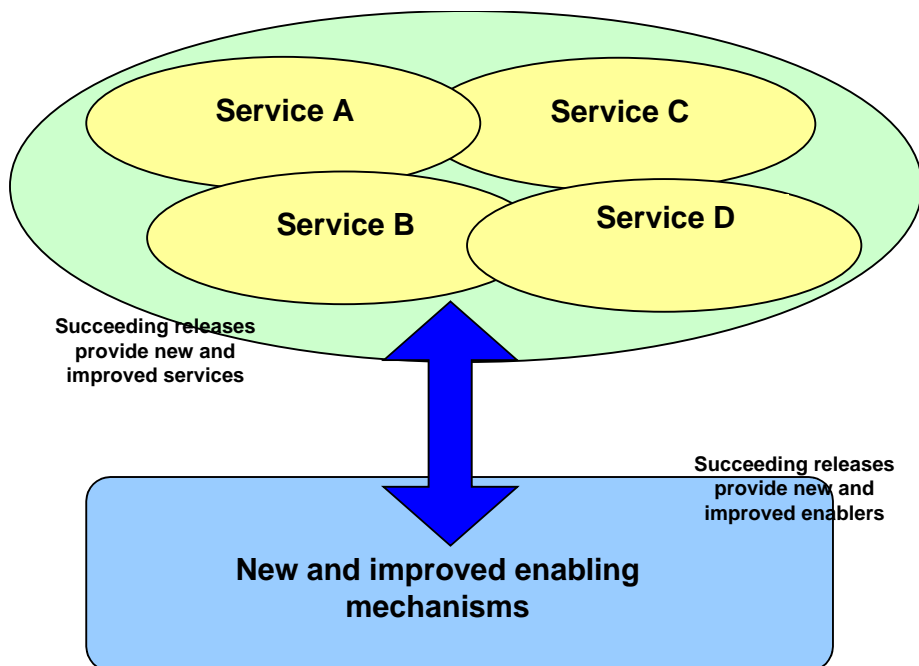
Figure 3: Succeeding releases provide new and improved services and enabling mechanisms

In Release 2000, new and improved enabling mechanisms and services may be made available. Additionally, in Release 2000 it shall be possible to offer services without using circuit switched network capabilities. In this case, the set of services available to the user, and the quality of the offered services shall be no less than that available in networks which use circuit switched enablers. This is shown in figure 4.
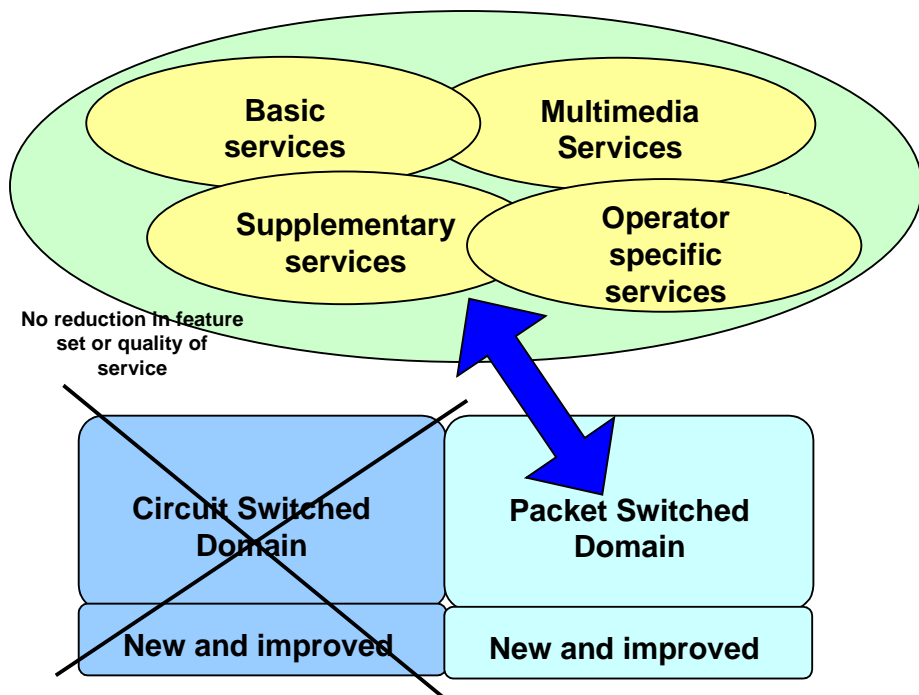
Figure 4: Network option

4.6   Simple classification of Release 2000 services.

Operators may offer a very rich set of services to their customers as shown in figure 5. Most of the services are related to difference types of telephony services, with various additions (e.g. VPN) and supplementary services (e.g. call forwarding). Additionally it is possible to offer non-call-related services like e-mail.

This chapter presents a simple classification of services from the 3GPP specifications point of view.

Basic services are divided into circuit teleservices (TS 22.003 [6]) and bearer services (TS 22.002 [15]) and they can utilise standardised supplementary services (TS 22.004 [7]). Basic services have not been changed much since they were introduced for GSM.

GPRS (22.060 [5]) provides IP bearer services. SMS, USSD and UUS can also be considered as a bearer service for some applications, however this had not been the original purpose.

IP multimedia services are the new services in Release 2000, including IP telephony. Supplementary services for IP multimedia services are not standardised but implemented using the toolkits or at the call control level. IP multimedia services use GPRS as a bearer.

Value added non-call related services include a large variety of different operator specific services. They are usually not specified by 3GPP. The services can be based on fully proprietary protocols or standardised protocols outside 3GPP.

In order to create or modify the above services (both call and non-call related services) operators may utilise toolkits standardised by 3GPP (such as CAMEL or LCS) or external solutions such as IP toolkit mechanisms. Pre-paid is an example of an application created with toolkits that may apply to all of the above services categories.
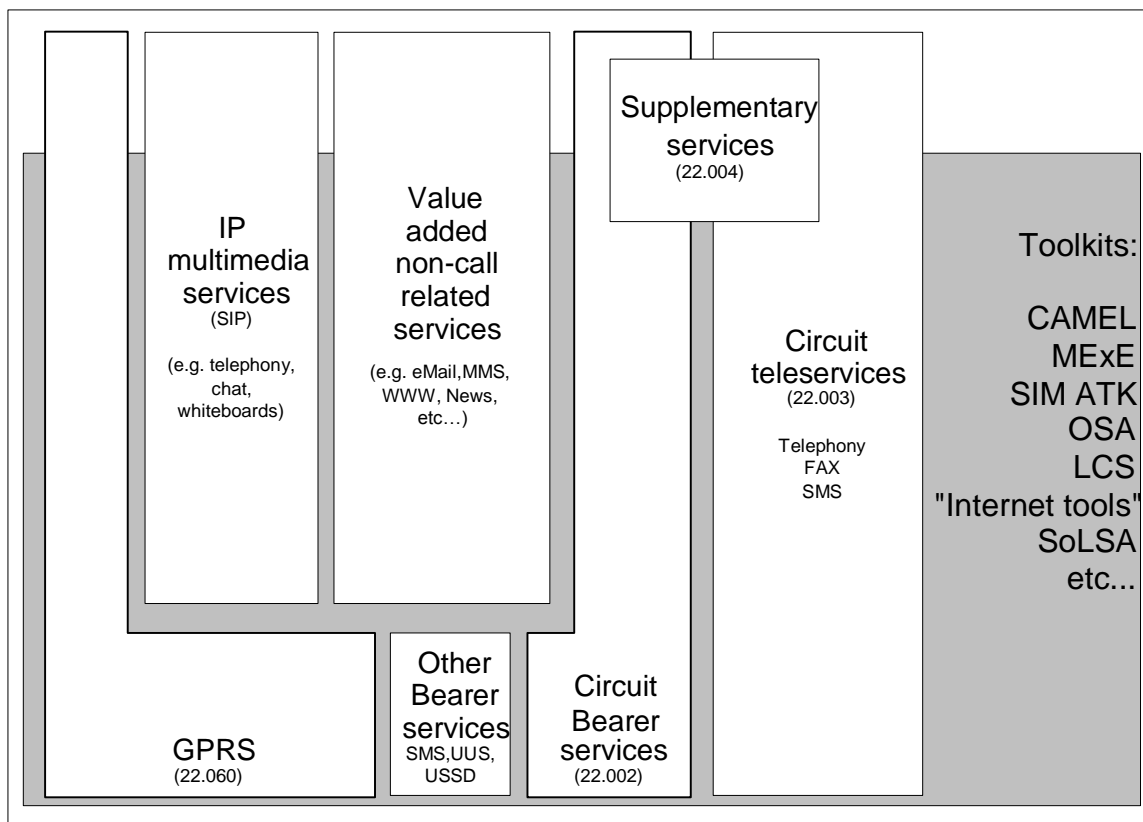
Figure 5: Service classification

4.7   Services, call control and transport

A Release 2000 network may have a circuit domain, a packet domain, or a hybrid network consisting of a circuit domain and packet domain network infrastructure. In addition to the basic services available from Release 99 new services, termed multimedia services, shall be available. Multimedia services may also enable enhanced usage and

management of basic services. The relationship between these basic and multimedia services, and the circuit/packet transport may be logically depicted as shown in Figure 6.
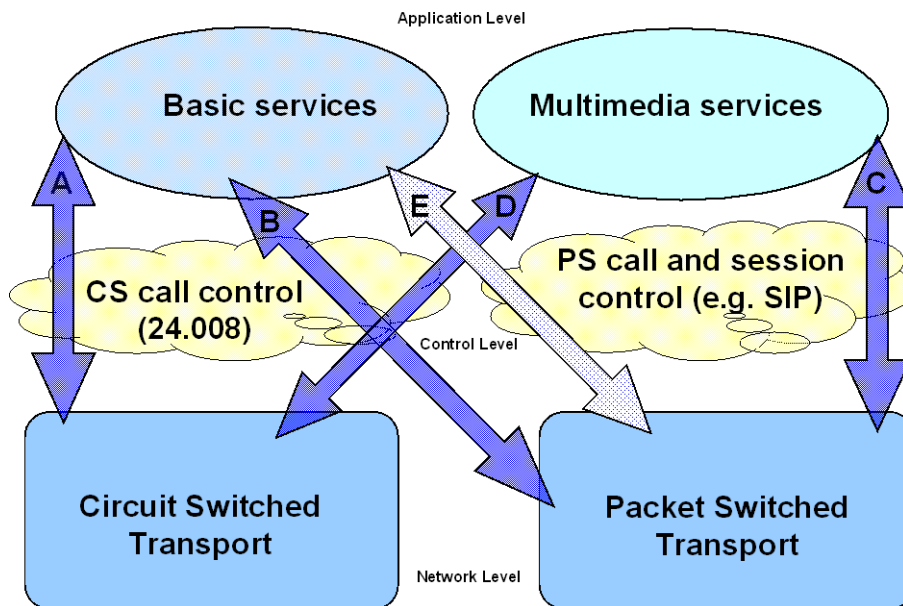
Figure 6: Services and domains

The logical relationship between the basic services, multimedia services, call control and the transport mechanism is subsequently described.

- "A" relationship
  The "A" relationship refers to the existing relationship between the basic services, circuit switched call control and circuit switched transport (such as existing GSM telephony).
- "B" relationship
  The "B" relationship refers to the support of basic services based on circuit switched control protocol and a packet switched routing and transport. 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 where a backward compatibility to existing CS-based voice terminals is provided.
- "C" relationship
  The "C" relationship refers to the relationship between the IP multimedia services, packet call and session control and the transport. The "C" relationship includes use of the existing capabilities of GPRS to provide IP multimedia services, and represent evolutionary enhancements to further the deployment of IP based services. The "C" relationship is not merely the evolution of the circuit switched services and mobile terminals to the IP multimedia services 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 built on the basic services of the "A" and "B" relationships across the "C" relationship, although some equivalent services (e.g. emergency services) are required. It is envisioned that in future releases which provide all services through a single service / transport relationship, that the network will enable provision of the same or similar services that users are accustomed to, with no reduction in quality of service, security, privacy, or authentication.

  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 shall provide sufficient capabilities to allow a Service Provider to develop and implement Release 2000 versions of these services that would have the same user interface 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.

An IM Service is typically based on one or more independent GPRS bearer services with defined QoS parameter. The network has limited knowledge (QoS parameters, type of IWF,...) on the type of application using the different data bearer services.

- "D" relationship
  The "D" relationship refers to the relationship between the multimedia services and circuit switched transport (e.g. H.324 supported in Release 99).

- "E" relationship (Not envisaged within Release 2000)
  The "E" relationship refers to the relationship of the existing basic services (including existing supplementary services built on the basic services) based on new packed switched control protocols, routing and transport (e.g. the re-implementation of 22.002 [15], 22.003 [6] and 22.004 [7] based on IP protocols). This is not envisaged within Release 2000.

4.8 High level service requirements

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

Release 2000 shall:-

1. in general, provide backwards compatibility with the services offered by Release 99 standard (including existing GPRS and circuit switched technology basic services, supplementary services, and operator specific services)
   a) the "C" relationship in Figure 6 shall provide sufficient capabilities to allow a Service Provider to develop and implement Release 99 circuit switched services as equivalent IP multimedia services that would be transparent to the end user. Further, Release 99 circuit switched services implemented across the "C" relationship may utilise the additional capabilities of the Release 2000 multimedia environment (e.g. existing and new service capabilities) to provide the end user with enhanced capabilities and improved user interfaces
   b) not all of the existing Release 99 circuit switched services need to be supported as IP multimedia services. Some exceptions may exist, as identified in the feature list in Annex A. A minimum set of IP multimedia services capabilities shall be defined to enable roaming
   c) enable service compatibility and access independence, it shall be possible to implement evolving IP multimedia services to be compatible with the same services when used via other types of accesses, e.g. via fixed lines
2. enable provision of IP multimedia services with the same (or greater) quality of service as their equivalent Release 99 GPRS and circuit switched services (if they exist)
   a) the enabling mechanisms (transport technology, etc.) shall be transparent to the user
   b) networks supporting IP multimedia services shall have the ability to provide, on an end to end basis, a Quality of Voice at least as good as that achieved by the Release 2000 circuit-switched (e.g. AMR codec based) wireless systems when interworking with other networks supporting IP multimedia services, other access networks, PLMNs or PSTNs
3. enable the possibility for IP multimedia services to be provided without a reduction in privacy, security, or authentication compared to corresponding Release 99 GPRS and circuit switched services
4. support roaming for IP multimedia services
5. enable the support of IP multimedia services using standardised capabilities or transparently over GPRS bearer services
6. respect spectrum efficiency (e.g. over the air interface, potential use of header stripping/compression techniques) for IP multimedia services
7. be able to support simultaneous realtime and non-realtime IP multimedia services
8. be able to support QoS parameters for realtime packet bearers
9. enable support of negotiable QoS for IP multimedia services both at the time of a connection setup as well as during the connection
10. be able to support interworking between the packet and circuit switched services, and with PSTN, ISDN and Internet
11. enable compliance with regional regulatory mandates for IP multimedia services (see Annex C).
12. enable support of O&M functionality (e.g. operations, administration, maintenance, provisioning etc.) by the IP capabilities of the network
13. enable support of charging and billing
14. enable support of flexible addressing schemes (e.g. for IP multimedia services users allocate a new number, re-apply current users' circuit switched numbers, different CS and PS domain numbers)
15. enable Release 99 UE to provide the Release 99 services to the subscriber when operating in Release 2000 networks

16. allow a Release 99 circuit switched UE to operate and use the services in a Release 2000 network configuration shown by Relationship B of figure 6
17. optimise the routing of user data and signalling between different networks to support services without quality degradation

4.9   IP Multimedia services high level requirements

The support of IP multimedia services in Release 2000 requires a flexible approach which will allow operators to differentiate their services in the market place as well customise them to meet specific user needs. This can be achieved through the use of standardised interfaces to service capabilities (e.g. MExE, SAT, CAMEL and OSA) in both networks and terminals, for the creation of services. This approach has already been adopted for the specification of the service capabilities provided by VHE (see [3], [4] and [8]).

For Release 2000, in general IP multimedia service descriptions shall only be used to determine the requirements for standardising service capabilities, and shall not result in the services themselves being standardised.

One of the key requirement for 3GPP is the standardisation of Service Capabilities to enable the support of IP multimedia services. See clause 7.2.

4.9.1 Basic requirements

The following basic requirements shall be supported for IP multimedia services:

1. IP multimedia services shall not just simply replicate the circuit services supported services, but enable the provision of a whole range of new services exploiting IP capabilities.
2. IP multimedia services shall be able to support the user with a consistent behaviour regardless of how they are accessed (e.g. menu on phone, WWW page from home or the office, or voice activated services, in the home network or in a serving network).
3. IP Multimedia services shall be able to personalise support of individual media components of a multimedia call in a user profile, and not necessarily handle all media components of a multimedia call in the same way. An example may be a multimedia call where the voice medium is forwarded to the user and the video medium forwarded to an appropriate terminal or storage device.
4. The user shall be able to personalise IP multimedia service behaviour at an "object" level within his service environment in a user profile. For example, the user may personalise forwarding for email, SMS message, MMS message, real time voice, real time multimedia, image, etc.
5. Call completion shall not simply be based on routing to a destination, but take into account the personalised media requirements of the involved parties (e.g. media splitting, media translation etc.). This will be subject to personalisation settings laid down in the User Profiles of the calling and called party.

4.9.2 User personalisation and administration of services

It shall be possible for the user to manage their IP multimedia call handling in various ways. The user shall be able to provision and activate services as follows:

1. For all subsequent calls
   - Via the terminal
   - Via a User Service Modification service on-line (voice or WWW)
   - When a call arrives or is in progress
2. For the next call
   - Via the terminal
   - When a call arrives or is in progress

For example, it shall be possible to change diversion and forwarding services criteria in a flexible way, from a range of terminals and the service to be consistently supported.

4.9.3 User context

It shall be able to support modified handling of an IP multimedia call as a result of the user context of the calling or called party. For example, for an incoming IP multimedia call the user context of the called party could be:-

- the called party's communications capabilities (i.e. the basic services of the called party)
- the called party's environment, which will be described by various parameters including:
  - time
  - location
  - terminal capabilities
  - circumstances (e.g. driving)
  - called party's identification (e.g. user at work, user at home etc.)
  - called party's choice of communication type (e.g. voice only)
  - called party's preferences, resulting in the call being:
    - presented to the called party
    - rejected subject to user specified criteria (e.g. calling party ID, media type, location)
    - diverted to another destination
    - diverted to a different medium (e.g. voice to email).
    - the called party's state, i.e. detached, idle mode, in a call, etc
- calling party's identification (e.g. work, family, anonymous)

The above example list of user context information is not exhaustive, and identifies user context information which may be managed by the user (e.g. in a user profile), or may be pertinent to the network. It shall be possible to perform handling of the context to process the multimedia call in one or more of the following:-

1. serving network,
2. home network
3. third party
4. calling party's terminal
5. called party's terminal.

Similar support shall be available to modify the handling of an outgoing multimedia call with respect to the context of the calling party.

4.9.4 Example multimedia scenarios

The following example scenarios describe the personalised handling of individual media in multimedia services (note that this list is neither complete nor exhaustive):-

1. The user is in a voice call, and receives an incoming video call. The user decides not to accept the call, but diverts the incoming video to a messaging system. Further, the user is given an indication that there is a video message in his mail box
2. The user is in a voice call, and receives an incoming video call. The user decides to accept the call but wishes to switch between the two calls.
3. The user is idle in a network and not involved in a call. The user modifies his user profile to divert all voice calls other than those from high priority, pre-identified callers (e.g. his boss). In this scenario all emails and text messages continue to be received regardless of the sender.
4. On receiving a call, the calling party's identity is displayed (if not restricted) and user shall be able to decide whether to accept the call, or divert to a messaging system. The user shall be able to request media handling of the call (e.g. media splitting to different destinations, media conversion).
5. The user is busy in a call when receiving an incoming call, but responds to the calling party that he will return the call later. The user may request that the calling party's details (if not restricted) are stored with a reminder in user's profile.

4.10 High level Quality of Service requirements

Support of QoS in Release 2000 is a means of supporting QoS network resources management, and negotiating the amount of network resources given to a user by the underlying network. The network may employ a number of QoS mechanisms, and the end user negotiates the network QoS through a set of standard parameters.

Note: the QoS parameters, which may be modified, require to be clarified

Release 2000 shall support the following QoS requirements:-

- The QoS shall be negotiated, both at call/session set-up and during the course of a call/session due to changes in network conditions or user requirements and /or application requirements. The QoS negotiations shall be based on user subscription profile , terminal and network capabilities and capacities

- Mobile terminals that only support GPRS QoS mechanisms shall require the network to act on its behalf in securing end to end QoS when required by the user in subscription information, as in this case there is no application layer that could request end to end QoS.

- Mechanisms whereby service providers can implement QoS based pricing. This implies that the network identifies the QoS requested and the QoS provided in the CDR.
- The network and the subscriber/application shall be able to dynamically negotiate the quality of service level for the call/session.

5.　Release 2000 services

Release 2000 development is based on an evolution of 3GPP Release 1999. Many items have already been commenced in earlier releases and will be improved or finalised for inclusion in Release 2000. All Release 99 services shall be supported in Release 2000, unless a specific service is deleted from Release 2000 as detailed below.

5.1　New Release 2000 services

This following major service developments are planned for Release 2000.

- IP multimedia services
- CAMEL phase 4 (see 22.078 [10]).
- MExE Release 2000 (see 22.057 [9])
- LCS with UTRAN (see 22.071 [16]).
- SoLSA with UTRAN (see 22.043 [17]).
- Wideband AMR speech codec (see 22.003 [6]).
- Global Text Telephony

In addition there are many enhancements to Release 99 services. The above list is neither exhaustive nor complete.

5.2　Release 99 services supported in Release 2000

All Release 99 services are included in Release 2000.

6　Virtual Home Environment (VHE)

The Virtual Home Environment (VHE) is defined as a concept for a personal service environment (PSE) portability across network boundaries and between terminals. The concept of the VHE is such that users are consistently presented with the same personalised set of features, user interface customisation and services in whatever network and whatever terminal (within the capabilities of the terminal and the network), wherever the user may be located [8].

6.1   Support of IP multimedia services

VHE shall be transparent to the transport mechanisms. New or extended service capabilities may be required to support IP multimedia services in Release 2000.

While not specified within Release 2000, the broad concept of VHE can also include IP applications or IP protocol used within the framework to provide subscribers with value added services.

6.2   Personal Service Management

Additions to Release 99 are required for Release 2000 personal service management.

6.3   Applicability of existing toolkits

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

Release 2000 shall incorporate improvements for VHE and OSA to support IP multimedia services, e.g. improvements to service capability features, service capability servers, user profile etc. This will give operators and 3rd party service developers the opportunity to create IP multimedia applications and services for Release 2000 networks.

Reuse of already implemented applications and services are also important. CAMEL, MExE, SAT and OSA, which are service capabilities in the VHE for Release 99 and will also be supported in Release 2000.

VHE/OSA Release 2000 shall include new (if required) and enhanced service capabilities to support IP multimedia services.

6.3.1 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 with both CS services and IP multimedia services. This shall 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 shall be maintained.

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

Operators shall be able to re-use their existing CAMEL services for IP multimedia services (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 packet switched and circuit switched domains in a seamless fashion.

CAMEL phase 4 shall include enhanced capabilities to support multimedia services. The enhancements may have an additional impact on the underlying network.

6.3.2 MExE

Release 2000 shall incorporate improvements made in MExE Release 2000 to support a new classmark and user profiles (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 shall require that applications, service logic and content downloaded in Release 99, may 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.

MExE Release 2000 shall include enhanced capabilities to support IP multimedia services.

6.3.3 USAT

Release 2000 USIM Application Toolkit (USAT) shall incorporate improvements made in USAT Release 99 (see [11]).

USAT Release 99 provides the ability for operators to download applications to USIMs from servers, and remotely control the content of USAT USIMs. SAT Release 99 also allows the Toolkit Applications to display additional menus to the user, for Toolkit related services, and allow call control of user dialled digits.

USAT terminals interact with the USIM using capability negotiation, and it shall be possible to continue usage of the capability negotiation for IP multimedia services.

Release 2000 shall include enhanced USAT capabilities to support IP multimedia services.

6.3.4 Open Service Architecture (OSA)

The Open Service Architecture is defined to enable applications and services to use service features of networks through an open interface. The applications and services may be implemented in standard programming languages (e.g. JAVA, C++) and on standard platforms.

6.4   Interoperability between toolkits

Any potential interoperatability issues between the toolkits require to be identified.

6.5   Fixed Mobile Convergence

VHE shall be access network independent. Requirements on how this is realised shall be specified in the Release 2000 specifications. Questions such as what level of VHE should be supported when a mobile terminal requires access from a fixed network and how will this be supported shall be considered in the Release 2000 VHE specification. To realise this level of support close collaboration with other ETSI groups in this area such as SPAN group is required.

7   Service continuity and new services

7.1 Relationship between services and subscriptions

The following figure provides a hierarchical model of relationship between subscribers, subscriptions and service availability in a mobile communication environment.
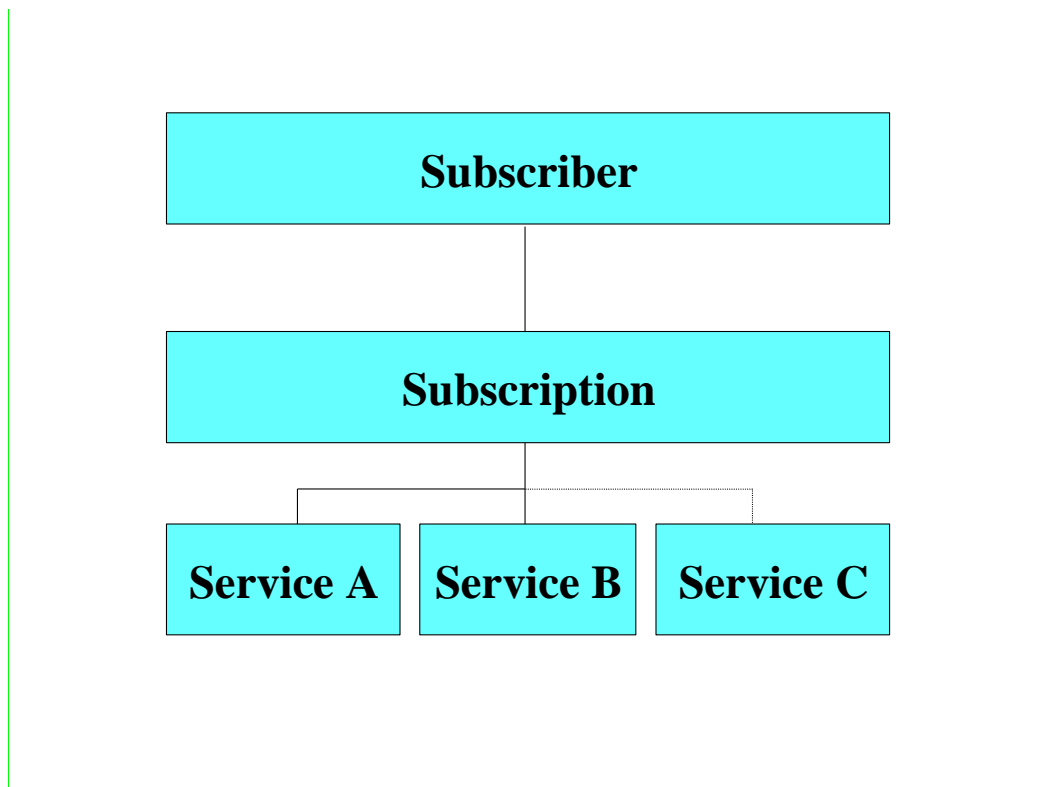
Figure 7: Subscriber, subscription and services relationship

In a typical usage scenario a subscriber has one subscription to a set of services (see figure 7) provided over a network with a combination of a CS Domain and/or a PS Domain and an IP multimedia subsystem (as an extension to the PS Domain). Within this subscription the subscriber has subscribed to one or more services belonging to the services classes such as Basic Services (e.g. Teleservices, Bearer services), GPRS and IM-Services (IP-based multimedia services). In addition to the GPRS subscription, information on the allowed QoS parameter ranges shall be contained in the GPRS service profile. In addition the subscription includes information on the supplementary services available to the subscriber.

In general it is a requirement to allow the use of independent services simultaneously (i.e. Basic, GPRS, IP multimedia and operator specific). It is required to have the flexibility to route calls (e.g. from PSTN using IP multimedia services or CS services) dynamically based on e.g. congestion, etc.

7.2 Requirements concerning service delivery

1. The network usage (HPLMN and VPLMN) shall be based on the service profiles within the subscription and roaming agreements between operators.
2. HPLMN shall be able to decide on the service delivery in a roaming scenario.
3. If an offered or required service (e.g. voice) could be provided with different technologies within the serving network, the serving network decides on the service delivery based on subscription information during or after the network selection procedure. The subscription information shall include preferences for service delivery controlled by the Home Operator and/or subscriber (e.g. if Circuit service for speech or IP multimedia service (including IP telephony).
4. Depending on the serving network conditions (e.g. load,...) at call set up this decision might be overridden within the limits indicated in the service profile.
5. In the case the service profile does not allow an alternative delivery and the requested delivery method is not available in the serving network the service will not be offered to the subscriber, a "not reachable" condition will be generated in the serving network at call set up and the call will be processed according the service configuration (e.g CFNR). This applies also to data bearer services with defined QoS parameters (or parameter ranges).
6.   If there is no preference information of the HPLMN available the decision on the delivery method made by the serving network is based only on the network conditions of the serving network.

Examples:

- An incoming voice call for a subscriber with a dual/multi mode terminal (e.g.UTRAN/GERAN) could be delivered in a Release 2000+ hybrid network as IP-telephony call or CS voice call (TS11). The delivery decision of the serving network is based on the preferences of service delivery within the service profile and the network conditions. If there is no preference information of the HPLMN available the decision is made only on the network conditions from the serving network.
- An incoming data service (e.g. GPRS with QoS for real time audio). The network can not provide the QoS at call setup. Both the originating and terminating application shall be informed about the possible QoS configuration for that call. The further handling (setup continuation, termination) depends on the decisions of the applications.

Note: Release 2000+ means Release 2000 and beyond.

7.3 Support of Release 99 services

Support of Release 99 services 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 continued support of such services in Release 2000. Existing Release 99 services shall as a principle also be supported in Release 2000, and any exception shall be explicitly identified.

7.4 Support of evolving IP multimedia services

It is important that Release 2000 supports evolving IP multimedia services and applications. The requirement for access independence of IP multimedia services implies, that if an IM service can also be accessed via other types of accesses (such as, e.g. fixed lines), it shall be possible to implement such a service in a way the end user experiences the same service behaviour irrespective of the access methods.

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

If a choice has to be made for Release 2000 IP multimedia services between being compatible either with IP based services or with existing GSM standardised supplementary services, then in principle the service compatibility shall be with the evolving IP multimedia services, however this must be evaluated case by case.

7.5 Support of supplementary services

New or operator specific supplementary services shall, as a principle, not be standardised, but instead be created using the VHE service capabilities (i.e. toolkits), as stated for Release 99 [3].

Release 2000 shall not specify in detail how Release 99 services are implemented, but solely identify the requirement for them to be supported (see Annex A).

To enable continued support of services, the existing Release 99 supplementary services in the CS domain shall be supported as specified in [7].

7.6 Roaming Requirements

This clause identifies the requirements for a mobile user to register and obtain services on the roamed-to network. Whether this is through use of the same terminal, SIM etc. is not discussed, with the point being that the home network subscription is used on the roamed-to network in some way.

The services available depend on the service profile and the required/allowed delivery method. Requirements for handling services that are not available in the serving network are the same as the ones defined in the service continuity tables in handover section.

Table 1 lists the required roaming scenarios:

Home Network	Roamed-to Network
3GPP Release 2000	3GPP Release 2000

3GPP Release 2000	Release 99 (incl. GPRS)
3GPP Release 2000	GERAN
3GPP Release 2000	ANSI-136 (incl. EGPRS)
Release 99 (incl. GPRS)	3GPP Release 2000
GERAN	3GPP Release 2000
ANSI-136 (incl. EGPRS)	3GPP Release 2000

Table 1: Roaming scenarios

Access to wired IP Networks with a mobile terminal, using a variety of access technologies (e.g. Bluetooth and Hiperlan) is likely to be viable in the near future. Therefore, consideration should be given in Release 2000 to issues such as authentication, registration and accounting, so that it does not preclude that in the future mobile subscribers can gain access to their IP multimedia services via wired IP network connections.

A number of issues have been identified which need to be dealt with in detail, these are as follows:-

- **Provision of Home Services**
  A User roaming to a visited PLMN (Release 2000) must be able to use services as provided in the home PLMN (Virtual Home Environment).
- **Home Environment awareness of roamed-to network capability**
  The home network might need to change the technical deliver mechanism according to the capability of the far end network in order to provide VHE. This is needed e.g. to ensure that Handling of Incoming Multimedia Calls when roaming in CS network are handled appropriately from the subscriber and operator point of view.
- **Supplementary Service Co-ordination between Release 2000 and legacy networks**
  Investigate the capabilities of the legacy services and try to map these capabilities between Release 2000 and legacy networks. Some IP multimedia services will not be supported in certain legacy networks.
- **Network selection**
  The time required to acquire the "correct" roamed-to network needs to be optimised. Especially if this involves selection based on the services offered (e.g. camp on one network, check services offered, skip to another network).
- **Service restriction**
  When a subscriber is subscribed to any services limiting the service offering like pre-paid, Barring SS, FDN or BDN the limitations must be fulfilled in roaming to network or if that cannot be provided the roaming shall be prevented.

7.7 Service continuity and handover requirements

In addition to the requirements identified in TS 22.129 Release 99 [14], the tables below define the service continuity and handover requirements for Release 2000. Supported end user QoS for IP Multimedia services is the same as defined in 22.105 [4] (clause 5.5).

Service continuity requires that a user's communication services (bearer- tele- or IM services) in a seamless way when moving within the serving network, restricted to the limitations (coverage limitation, capacity limitation, ...) of the network. In general Service continuity is supported by handover or equivalent procedures.

Additionally, if a serving network for whatever reason is no longer able to provide the currently invoked tele- or IM-services over the currently assigned bearer(s) but can provide equivalent services (fully or partially) over a different bearer to the user, it shall support the transition of the original tele- or IM-service to the equivalent service. This transition shall proceed in such a way that a user is provided with the experience of a service (e.g. voice) being continued.

In the following sections "equivalent" services will be explicitly indicated.

For the scenarios outlined in the tables below, the specifications for Release 2000 shall cover both the intra-PLMN and the full inter-PLMN cases.

7.7.1 Service continuity between CS services and IM Services

Service continuity required?		To CS services		To IM Services		To CS ANSI 136 basic services
		UTRAN	GERAN	UTRAN	GERAN	
From CS Services	UTRAN	Yes note 7	Yes Note 7	Yes (note 1)	Yes (notes 1,3)	Note 4
	GERAN	Yes note 5	Yes Note 5	Yes (note 1)	Yes (notes 1,3)	Note 4
From IM services	UTRAN	No	Yes (note 2)	Yes	Yes (note 3)	Note 4
	GERAN	No	Yes (note 2)	Yes (note 3)	Yes (note 3)	Note 4
From CS ANSI 136 basic services		Note 4	Note 4	Note 4	Note 4	Out of scope

Table 2: Service continuity between CS Services and IM Services

The requirements for CS Multimedia services:

Service continuity is required from CS Multimedia services (UTRAN and GERAN) to CS Multimedia services (UTRAN and GERAN).

Work is currently ongoing in CN3 on service continuity for fallback and swap from multimedia to speech.

The following notes apply to table 2:

Note 1: Seamless service continuity of the voice medium of a voice call to an IP multimedia call is required.
Note 2: Seamless service continuity of the voice medium of an IP multimedia call to a CS call is required.
Note 3: Service continuity of IM Services is not applicable to and from GERAN Release 99 as the IM Services service classes are not supported by GERAN Release 99. Note that the "voice-only" service is seen as a special case of the IP Multimedia service.
Note 4: Requirements dependent on the result of the associated work item (UMTS Interworking with ANSI-41 networks).
Note 5: Same requirements as in Release 99 (c.f. 22.129) [14].

7.7.2 Service continuity of GPRS

For service continuity of GPRS sessions, the requirements for Release 1999 are maintained (cf 22.129 [14]. In addition, service continuity is required between GERAN Release 2000 and UTRAN, and with GERAN Release 2000.

Note: Service continuity for conversational/streaming/interactive GPRS sessions is not applicable to and from GERAN Release 99, due to the lack of support for the conversational/streaming/interactive GPRS within GERAN Release 99.Although the QoS required may not be available in the network to which the call has been handed over to, the bearer shall be maintained as long as possible. The user makes the decision whether to continue with the call based on the application behaviour.

7.7.3 Service continuity between (CS Services and IM services) and GPRS

Service continuity is not applicable in the following scenarios:

- between CS Services and GPRS
- between IM Services and GPRS.

7.7.4 GERAN/UTRAN handover performance requirements for IP multimedia services and GPRS

The GERAN/UTRAN handover performance requirements for IP multimedia services and GPRS are defined in table 3 below.

Service Classes	Application	Degree of symmetry	Data rate	Key performance parameters and target values			
				Max. service interruption (Audio/Video) mute)	User Data Transmission Requirement		
					Max. data trans. Delay	Max. service impact	Information loss
Conver-sational Class	Telephony	Two-way	4-25 kb/s	<=150ms	TBD	NA	TBD
Streaming Class	Streaming audio	One-way	32-128 kb/s	TBD	TBD	NA	TBD
	Streaming video	One or two way	32-384 kb/s	TBD	TBD	NA	TBD
	Bulk data transfer/retrieval	One-way		NA	TBD	TBD	None
Interactive Class	Interactive best effort data	Two-way		TBD	TBD	TBD	None
Background Class	Best effort data	One-way		NA	TBD	TBD	None

Table 3: GERAN/UTRAN Service continuity performance requirements for IP multimedia and GPRS

TBD = To Be Determined. It is very important to get these TBD replaced with true value as soon as possible
NA = Not Applicable
None = Not allowed

7.7.5 Justification for service continuity requirements

An example service scenario is given below for each requirement in the tables above to justify that requirement.

From CS services (UTRAN/GERAN) to CS services (UTRAN/GERAN): Continuity of CS services when moving (i) within areas of UTRAN coverage, (ii) within areas of GERAN coverage, (iii) from and to islands of UTRAN coverage amongst widespread GERAN coverage.

From CS services to IM services: Service continuity of a CS call to IM Services to allow use of IM services, (for example if the CS call started in widespread GERAN Release 99 coverage and the user moved into UTRAN/GERAN Release 2000 coverage).

Note: Concern was expressed at the complexity of this handover scenario. The difficulties of supporting such a scenario and the alternative of an application re-establishing the multimedia call as a CS call were proposed. The feasibility of easily upgrading existing GERAN coverage to support QoS aware IP services could be studied. Further work is required to determine which components (and to what extent) are handed over i.e.:
- does a video call degrade to a voice-only call?,

- which degradation rules should be applied?,
- what delays are acceptable?,
- which bearers continue to be supported?,
- is original control function maintained?.

From IM Services (UTRAN/GERAN) to CS services (GERAN): Seamless Service continuity when moving from islands of UTRAN/GERAN Release 2000 coverage to more widespread GERAN Release 99 coverage.

Note: See note for " From CS services to IM services"

From IM Services to IM Services: Continuity of IM Services when moving (i) within areas of UTRAN coverage, (ii) within areas of GERAN coverage, (iii) from and to islands of UTRAN coverage amongst widespread GERAN coverage.

From conversational/streaming/interactive GPRS to conversational/streaming/interactive GPRS: Continuity of GPRS when moving (i) within areas of UTRAN coverage, (ii) within areas of GERAN coverage, (iii) from and to islands of UTRAN coverage amongst widespread GERAN coverage.

From GPRS background to GPRS background: Continuity of calls when moving (i) within areas of UTRAN coverage, (ii) within areas of GERAN coverage, (iii) from and to islands of UTRAN coverage amongst widespread GERAN coverage.

From conversational/streaming/interactive GPRS (UTRAN/GERAN) to background GPRS (GERAN): Requirement to maintain a GPRS bearer when moving from islands of UTRAN or GERAN Release 2000 coverage to more widespread GERAN Release 99 coverage.

7.8 Naming, numbering and addressing

Release 2000 support of naming, numbering and addressing shall be compatible with those being developed for fixed, mobile and IP networks, and depends on the outcome from other related organisations (e.g. ETSI, Tiphon, IETF, and ITU-T). Also, developments in bodies such as the IETF and ICANN covering issues such as new Top-Level Domain names or the introduction of IPv6 will impact Release 2000. Given these interdependencies it is important that Release 2000 networks are capable of being flexible.

7.8.1 Names and addresses for IP multimedia services

Although naming and addressing solutions for Release 2000 shall depend heavily on the solutions produced in other fora, the following sections describe the top level principles,

7.8.1.1 Names

A variety of Internet names should be supported, including URLs, e-mail addresses etc.

Note: a clarification on the types of supported internet names requires further elaboration.

It is assumed that Release 2000 services shall be required to interwork with the PSTN, GSM and other networks. To enable this backward compatibility, for users to be called from the PSTN, Release 2000 users need to be reachable via E.164 numbers. The types of numbers will be many and varied and may depend on the local environment (e.g. a new E.164 number may be allocated for IP multimedia services). It is important that Release 2000 does not limit future capability by deciding on the use of a specific type of number. The following types of E.164 numbers shall be supported by Release 2000:

1. Geographic areas: CC + N(S)N
2. Global services: CC + GSN
3. Networks : CC + IC+SN

An objective is to move to future developments such as that being progressed in the ETSI STF 157.

7.8.1.2 Name and number portability

It is desirable to support name and number portability for IP multimedia services, however some aspects of this portability are outside the scope of this TR. The exact portability scenarios require to be identified.

Note: Although IP supports a degree of portability (i.e. an e-mail name can be used at various IP addresses) it is not standard practice to support full name portability from one provider to another. For example, it is not standard practice to port Joe.Bloggs@netA.com to a new provider NetB without changing the e-mail name to Joe.Bloggs@netB.co.uk, although some content providers do offer a 'name independent' service as a special service. In future this type of service could become standard practice with the 'name provider' being independent from the network provider. Networks must track developments in this area and be ready to implement flexible solutions.

7.8.1.3 Addressing

Release 2000 networks shall be capable of using IP addresses for routing. Release 2000 shall allow a mechanism to communicate with both IPv4 and IPv6 services.

GPRS shall continue to support the allocation of both static and dynamic IP addresses.

8   Service capabilities to support existing and new services

This subclause investigates the support of Release 99 services as equivalent IP multimedia services. The aim is to identify areas subject to standardisation in the Release 2000 specifications. Existing services are investigated in order to determine to what extent they can be supported by the Release 2000 toolkits without additional specification work. As a reference the appearance of the existing services to a user is taken.

This clause does not attempt to compare the IP multimedia service capabilities against as yet unidentified Release 99 additional features, however, any value added to existing services shall not be prevented and is not treated within this subclause. Extensions to the range of circuit switched services may also be considered for Release 2000, however they are not considered in this clause either.

One of the most important requirements for the control of IP multimedia services is that the necessary protocol actions are performed using functional peer-to-peer signalling. The protocol has to be designed in such a manner that the functional entities communicate with their peer entities (servers, clients or gateways (in case of interworking)) directly without assuming any network intervention.

Other requirements are:-

-    use of an open, standardised, extendible protocol for service control to enable service differentiation in multi-vendor networks
-    independence from vendor specific solutions and platforms
-    support of interworking with public networks (e.g. ISDN networks)
-    standardised and extendible service capabilities instead of standardised services
-    support of client/server service models.

The above list is neither exhaustive nor complete.

Services in Release 2000 can be created from the existing service capabilities (CAMEL, MExE, OSA and SAT) enhancements with possible and/or IP applications in the terminal and network.

The following options shall be available in the standards to enable service delivery in the new architecture:

- capability toolkits enhanced to control IP multimedia services, which will allow applications to be deployed in a vendor independent manner

- a framework that enables toolkits not standardised by 3GPP to be used to deliver services (e.g. adoption of IP recommendations to facilitate the IP applications)

- registration mechanisms which allow the network to understand the limitations of the mobile and thereby take appropriate actions.

Note: there is a concern that with a large variety of toolkits to create services, service interworking between terminals and networks may be compromised and needs to be addressed.

8.1 Possible realisation of existing Supplementary Services

By analysing and categorising existing Release 99 supplementary services this clause concludes that these services shall not be re-standardised, and identifies the new service capabilities which shall be standardised in order to create these services.

This clause:-

1. groups the main Release 99 supplementary services into several main categories of services
2. analyses each main category of services
3. identifies which services can be provided without standardisation by using the standardised service capabilities, given that these capabilities are enhanced to control IP Multimedia services (CAMEL, MExE, SAT and OSA toolkits)
4. identifies which functions (such as authentication, CAMEL call triggers etc.) shall require to be standardised

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

The following subclauses are not a complete list of supplementary services, which is elaborated in the feature list.

8.2 Basic Call

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

8.3 Barring Services

This category of services performs two basic functions:-

*   limiting the user's usage of subscribed services (e.g. no roaming, no long distance, session barring, bearer barring, QoS etc.)
*   bar calls to control the cost of the calls
*   bar incoming calls
*   bar outgoing calls

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

Specifying which services are available to users can be done through WAP, MExE, SAT, CAMEL and OSA toolkits. Outgoing barring services for basic services can be implemented using SIM Toolkit applications based on the number dialled, CAMEL and HLR (e.g. serving network applications), and MExE applications (e.g. MExE services, WAP WTA applications etc.). Incoming call barring services for basic services can be implemented using the network (e.g. CAMEL and HLR), and MExE applications (e.g. MExE services, WAP WTA applications etc.). Operator Determined Barring may also be supported using these toolkits. The barring of specific basic services with toolkits requires further investigation. Therefore, by using service capabilities there is no requirement for a specific call barring service to be standardised in Release 2000.

8.4 Advice of Charge

A feature in 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). Charging related information may also be made possible to users in advance of the call being made, in order to give an indication of the potential call costs.

Therefore the advice of charge may originate from the (mostly IN-based) charging services and delivered to the user by various means. Displaying of the received charging information to the user could be with MExE, as well as re-using capabilities to transport the Advice of Charge Information to the terminal and executing an AoC application in the terminal (as defined in 22.024 [18]). Different methods may also exist.

8.5   Call Forwarding

This category of services include immediate call forwarding, call forwarding on no reply, call forwarding on not reachable, call forwarding on busy, call forwarding on user initiation and call completion to busy subscriber. Further, the diversion may also be applied depending on the type of service (e.g. speech, data, multimedia, by media type etc.).

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

1.   immediate call forwarding per medium type
2.   call forwarding on no reply per medium type
3.   call forwarding on not reachable per medium type
4.   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 forwarding features other than those proposed above are required to be standardised.

This allows terminals to be capable of providing the call forwarding features when reachable (with the service logic securely downloaded using MExE, building on WAP's WTA where available), and the home network/serving network to handle call forwarding when the terminal is unreachable (using HLR/CAMEL).

The basic primitives may also provide the capability to offer call waiting, hold and transfer features through the MExE toolkit. For the case where the terminal is reachable, the selection of the terminal or the network to process the call forwarding may be an option.

8.6   Conferencing

Support for a service similar to the CS domain Multiparty service shall be required as an IP multimedia service.

As a new enhancement, 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 multimedia service shall allow multicast connections to be made outside of the cellular part of the core network.

In conclusion, the Multiparty service feature as an IP multimedia service where calls may consist of multiple media each possibly requiring specific handling) may at best be difficult to provide. It is therefore recommended that the circuit switched Multiparty service is supported by a new IP multimedia Conferencing service in the PS domain.

8.7   Pre-paid Service

The pre-paid service allows the subscriber to pay for telecommunication services prior to usage.

A pre-paid service subscriber establishes an account with the service provider to access telecommunications services in the home and roamed-to networks. Charges for telecommunication services are applied to the pre-paid service account by decrementing the account in real time. The pre-paid service subscriber may be notified about the account information at the beginning, during, or at the end of the telecommunications service. When the account balance is low the subscriber may be notified so that the subscriber may refill the account. When the account balance is below a pre-defined threshold, the subscriber's use of telecommunications services may be barred.

The support required from the toolkits to support the pre-paid service needs to be identified.

8.8 Service provisioning

The range of new IP multimedia services will require provisioning and configuration by users and service providers. Since the range of services and the services themselves are not standardised, 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.

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 SIP. This also means that 3GPP shall not standardise any mobile specific variants of these standards in 3GPP.

There are cases where today's IP based standards have to be modified to suit the mobile environment of Release 2000. In such cases, the relevant IP standardisation fora shall enhance ing those IP standards. 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.

10 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	February 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.

Annex A   PS Domain feature list evaluation for release 2000

Key to Table

E = Essential for release 2000, launch of commercial IP multimedia services is not viable with these missing or required in Release 2000 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 (as may be enhanced in Release 2000 or Release 2001)

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)

Where multiple selections are made for a feature, then no strong choice is made.

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
IP multimedia services							
IP telephony	Single medium IP voice call (using SIP) with end user perceived quality equal or better than GSM voice call		X				It must be possible for the MMI to 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		X				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).

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Emergency Voice Call	Basic emergency voice call over IP		X				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)			X	X		Service requirements FFS
Cell Broadcast 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 messaging (MMS)	Support of multimedia messaging in PS domain	X					Currently supported in R99 for MS/MS. Need to consider messaging to and from other access.
Text Telephony	IP calls capable of supporting realtime text conversation		X				FCC regulatory requirements are considered to be essential for Release 2000. Global Text Telephony (GTT) is one means of supporting this requirement.
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.	X					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.	X					Based on T.38.

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
GPRS							
Point-to-Point	As in 22.060	X					No additional standardisation work required for S1
Point-to-Multipoint	IP Multicast	X					Implementation of PTM services for example as defined within GPRS specifications
Point-to-Multipoint	PTM Group Call			X			Implementation of PTM services for example as defined within GPRS specifications
Point-to-Multipoint	PTM services such as PTM-Multicast, , IP Distribution Services (Multimedia Distribution Service)			X			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
Support of QoS mechanisms for real time services		?	X				QoS is also part of R99, but the features for real time conversational services might be delayed
Multi-session 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

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
IPv6 interworking		X					Same reqs as Rel 99 GPRS(?)
Speech to/from PSTN / ISDN / CS mobile / Release 2000 CS mobile			X				
IP Multimedia to/from Internet			X				Incl. Intranet.
Modem and ISDN interworking	Access to PSTN / ISDN dial up –data services with GPRS					X	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			X			Interworking standardised outside 3GPP.
Interworking with other access networks (e.g. cable)				X			
Interworking with intranets (including VPNs)	VPN functionality (firewall bypass) shall be supported	X					Already specified in 29.061. May possibly be supported at the application layer.
Roaming							Refer to the Roaming clause of this TR.
Handover							Refer to the Service continuity clause of this TR.

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
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.		X				Also includes incoming multimedia calls and barring when roaming.
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.		X				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.			X			
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.			X			

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Multimedia Call Holding	Allows served mobile subscriber to interrupt communication on an existing active multimedia call and then subsequently re-establish communication		X				
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.		X				
Advice of Charge	Supply user sufficient information to allow real-time estimate of the call charge.			X			
Caller Identification and restriction	Similar to CLIP, CLIR, CNAP and CNAR.		X				Could include additional IP related information such as IP address. Essential due to European, FCC and other regulatory requirements.
Connected Line Identification and restriction	Similar to COLP and COLR		X				Could include additional IP related information such as IP address. Essential due to European, FCC and other regulatory requirements.
Multimedia conferencing	Similar to Multiparty but is applicable to multimedia calls.		X	X			

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
Multimedia call-back when free / CCBS					X		
Closed user group (CUG) / community of interest					X		
Precedence and Pre-emption service					X		
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.	?	X				
CAMEL Support for Multimedia Services	Provides mechanisms to support multimedia services consistently & independently of the serving network.		X				CAMEL enhancements, but additionally needs to support multimedia calls. Implementation is FFS.
(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.		X				
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		X				e.g. SIP CSCF.

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
LCS for GPRS	Support of LCS on the PS domain required to meet regulatory and commercial requirements (e.g., 3GPP 22.071)		X				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).			X			
Lawful Surveillance / Intercept			X				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.		X				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.		X				
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			X			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. Work on this feature should wait until the results of the ETSI STF on Personal Identifiers for the 21st century are available.

Feature Name	Short description	R99	E R00	D R00	R01+	No	Comments/Notes
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.		X				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.		X				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		X				
Event/transaction based charging mechanisms (e.g. content based)	Definition of charging mechanisms for IP Services.		X				
Charging aspects – need to charge for each PDP context (PS sessions) independently		X					

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Annex B   Release 2000 document structure

Release 2000 introduces a set of new requirements and service descriptions mainly related to IP multimedia telephony. The IP based architecture offers an extended set of services, and may support services differentiating from services offered by circuit switched network.

Additional new specifications for IP multimedia services should be created when needed.

The following table presents the TSG-S WG1 document structure, with initial comments on their applicability for Release 2000.

New	Name	Release 2000 notes
22.001	Principles of Circuit Telecommunication Services Supported by a Public Land Mobile Network(PLMN)	Not relevant for PS Domain.
22.002	Circuit Bearer Services Supported by a PLMN	Not relevant for PS Domain.
22.003	Circuit Teleservices supported by a PLMN	Not relevant for PS Domain.
22.004	General on Supplementary Services	Not relevant for PS Domain.
22.011	Service Accessibility	No major changes seen.
22.016	International Mobile Equipment Identities (IMEI)	Relevant for both CS and PS domains. No major changes seen.
22.024	Description of Charge Advice Information (CAI)	See 22.004
22.030	Man-Machine Interface (MMI) of the Mobile Station (MS)	FFS
22.034	High Speed Circuit Switched Data (HSCSD) – Stage 1	Not relevant for PS Domain.
22.038	SIM application toolkit (SAT); Stage 1	Should be applicable to both sides as part of VHE. Further study for IP multimedia telephony needed.
22.041	Operator Determined Call Barring	Only applicable to CS domain. Further study for rel 2000 needed.
22.042	Network Identity and Time Zone (NITZ), stage 1	Only applicable to CS domain. Further study for rel 2000 needed.
22.043	Support of Localised Service Area (SoLSA) - Stage 1	Relevant for both CS and PS domains. Further study for rel 2000 needed.
22.057	Mobile Station Application Execution Environment (MExE); Stage 1	Should be applicable to both sides as part of VHE. Further study for IP multimedia telephony needed.
22.060	General Packet Radio Service (GPRS); Stage 1	Not relevant for CS domain. Detailed review needed.
22.066	Support of Mobile Number Portability (MNP); Stage 1	IP multimedia telephony related changes FFS.
22.067	Priority Set-up Service(PSUS); Stage 1(ASCI spec)	See 22.004
22.071	Location Services (LCS); Stage 1	Applicable to both domains, GPRS status not clear . Further study for rel 2000 needed.
22.072	Call Deflection (CD); Stage 1	See 22.004
22.078	CAMEL; Stage 1	IP multimedia telephony support need to be studied.
22.079	Support of Optimal Routing; Stage 1	Only applicable to CS domain. Similar requirement needed for rel 2000.
22.081	Line Identification Supplementary Services; Stage 1	See 22.004

New	Name	Release 2000 notes
22.082	Call Forwarding (CF) Supplementary Services; Stage 1	See 22.004
22.083	Call Waiting (CW) and Call Hold (HOLD) Supplementary Services; Stage 1	See 22.004
22.084	MultiParty (MPTY) Supplementary Service; Stage 1	See 22.004
22.085	Closed User Group (CUG) Supplementary Services; Stage 1	See 22.004
22.086	Advice of Charge (AoC) Supplementary Services; Stage 1	See 22.004
22.087	User-to-user signalling (UUS); Stage 1	See 22.004
22.088	Call Barring (CB) Supplementary Services; Stage 1	See 22.004
22.090	Unstructured Supplementary Service Data (USSD); Stage 1	See 22.004
22.091	Explicit Call Transfer (ECT) Supplementary Service; Stage 1	See 22.004
22.093	Call Completion to Busy Subscriber (CCBS); Stage 1	See 22.004
22.094	Follow Me; Stage 1	See 22.004
22.096	Calling Name Presentation (CNAP); Stage 1 (T1P1)	See 22.004
22.097	Multiple Subscriber Profile (MSP); Stage 1	PS domain FFS.
22.100	UMTS phase 1 capabilities	Not applicable to rel 2000.
22.101	UMTS service principles	Applicable to both domains. No major changes seen. Introduction to IP multimedia services and IP multimedia telephony might be good.
22.105	Services and service capabilities	Applicable to both domains. IP multimedia services are FFS
22.115	Service aspects: charging and billing	Applicable to both domains. IP multimedia services are FFS
22.121	Virtual home environment	Applicable to both domains. IP multimedia services are FFS
22.129	Handover requirements between UMTS and GSM or other radio systems	Applicable to both domains. IP multimedia services are FFS
22.135	Multicall	Applicable to CS domain only. GERAN applicability studied with EDGE.
22.140	Multimedia Messaging Service Stage 1	Applicable to both domains. IP multimedia services are FFS
22.960	Mobile multimedia services including mobile intranet and internet services	No impact identified.
22.971	Automatic establishment of roaming relations	No impact identified.
22.975	Advanced addressing	No impact identified.
22.976	Study on PS domain services and capabilities	Basis for S1 rel 2000 IP multimedia services work.

Annex C　　Regional Regulatory Requirements

Release 2000 shall enable compliance for IP multimedia services with regional regulatory mandates. The following list is neither exhaustive nor complete, and requires further elaboration in this annex.

US regional regulatory mandates services may include (and are not limited to):-
- emergency services (e.g. in the U.S., the FCC Phase 2 Emergency Services mandate which requires provision of geographic location information).
- lawfully authorized electronic surveillance (LAES)
- number portability
- text telephone services (e.g. TTY in the U.S.)
- priority services for Critical Communications (e.g. ITU-T Study Group 2 recommendation E.ieps (International Emergency Preference Scheme to support the need for telecommunications among essential users of public telecommunications networks in crises situations)).

Annex D　　　Case study of services realisation

D.1　　Case study on the support of Call Forwarding services

This category of services include any kind of call forwarding services (e.g. immediate call forwarding, call forwarding on no reply, call forwarding on not reachable, call forwarding on busy, call deflection etc.).

D.1.1　　Requirements

The main requirements of this category of services are:

- provide the functionality to re-route the entire incoming call, or individual media of the incoming call
- allow the re-routing on defined criteria (e.g. context of called party or calling party, time, location, call set-up indication, user indication, service type, etc.)
- be able to differentiate the service type of the incoming call (e.g. speech, data, multimedia, etc.)
- provide sufficient capabilities to select where re-routing shall take place (e.g. in the network, in a third party, in the terminal etc.)
- user controlled administration, activation, de-activation and interrogation from different points (e.g. via the terminal, in the network, via a third party, user's web-based service personalisation etc.)
- the ability to inform the involved parties (calling, served, forwarded-to) that call forwarding was invoked.

D.1.2　　Needed service capabilities

3GPP defines the service capabilities on which the services shall be built rather than defining the services themselves. This section identifies the functional support required from service capabilities which shall then be combined to build the required service.

The following functional support shall be provided by the service capabilities to fulfil the above defined requirements for the call forwarding services:-

- monitor for an incoming call
- provide user and/or terminal status (e.g. context of the calling or called parties)
- create a call to a (forwarded-to) destination for one or more media of an incoming call
- connect one or more media of an incoming call to one or more media of a (forwarded-to) destination
- identification of the service type of the incoming call
- administration (e.g. settings, modifications, interrogations, etc.) of the service data (e.g. forwarded-to destinations, trigger criteria, activation state, etc.) in the network, in a third party, in the terminal etc.
- definition of the functionality (service logic) for the execution of the service
- inform the involved parties about the invocation of a service.

D.1.3　　Functional description

In traditional telephony a user is provided with only a limited number of forwarding services (e.g. unconditional, in case of busy, etc.). With the introduction of service capabilities it shall be possible for forwarding services to be made dependent on a variety of conditions. Examples for these conditions are the state of the called party (e.g. absent, busy, no reply, location, context etc.), the kind of service type (e.g. speech, multimedia, data etc.), the caller identification, the time of day or day of week etc.

For each of the above mentioned scenarios it is required that the user can program the forwarding of incoming calls (or media of incoming calls) to different destination addresses. It shall be possible that the programming of the destination can be done locally at the home client (terminal, gatekeeper, proxy server), or by remote programming via connection to the home client (e.g. via WAP/Web access).

The following different possible locations for the execution of the service (service logic) may be considered:

- the **client** (terminal, server/gatekeeper) maintains the states of the calls it is handling. Therefore many of the services, like call forwarding are suitable for implementation in a client. Services which are personalised/customised or interact with the user to the user's own specific requirements (e.g. calendar etc.) are best implemented in a client. The user may control the call handling requirements for the cases when he is available, and by defining his forwarding requirements as part of his profile he may also control his call handling

requirements when he is unavailable. Such a profile could, for example, be webpage-based allowing fine-grain handling of each medium within his multimedia call. Further, the user's personalisation may also be accessed by the application/feature servers to provide continuity of service when the client is unavailable.

- an **application/feature server** implements services that are not suitable for client implementation (e.g. when the client is not reachable), but in general it is also be possible to implement many services at application/feature servers. For example services which interfaces with a group of clients (e.g. with the first available agent with a specific skill) are maybe best implemented in a server.

This application/feature server can also be used as a proxy or secondary client for those clients that are non-operational (e.g. powered-down, not registered, not reachable). Upon detection of the operational failure of a primary client, the gatekeeper notifies and routes all calls destined for the non-operational client to the server, which could then provide such services as call forwarding and messaging (for voice, facsimile and email).

A special case of such an application/feature server could be a **conference server** which maintains also the state of the calls that it is handling and which can provide n-way conferencing services.

D.1.4 Call flows

The following two examples show how a call forwarding service could be realised as an IP multimedia service. The message names used only indicate the function behind the signalling (the mapping of this functional flows to SIP messages is outside of the scope of this document).

The service capabilities supporting the function (e.g. usage of CAMEL, MExE, SAT, OSA), as well as the functional entity in which the service logic is executed (e.g. application/feature server, client) are not identified in this study. The "served party" entities in figure 6 and Figure 7 represent the network and/or the terminal capabilities. The box "An application…" in Figure 6 and Figure 7 depicts service logic which is processing the call, and information on the calling/called party's context is potentially obtained from several sources (e.g. calling/called party, the network, user profile, user's webpage etc.).
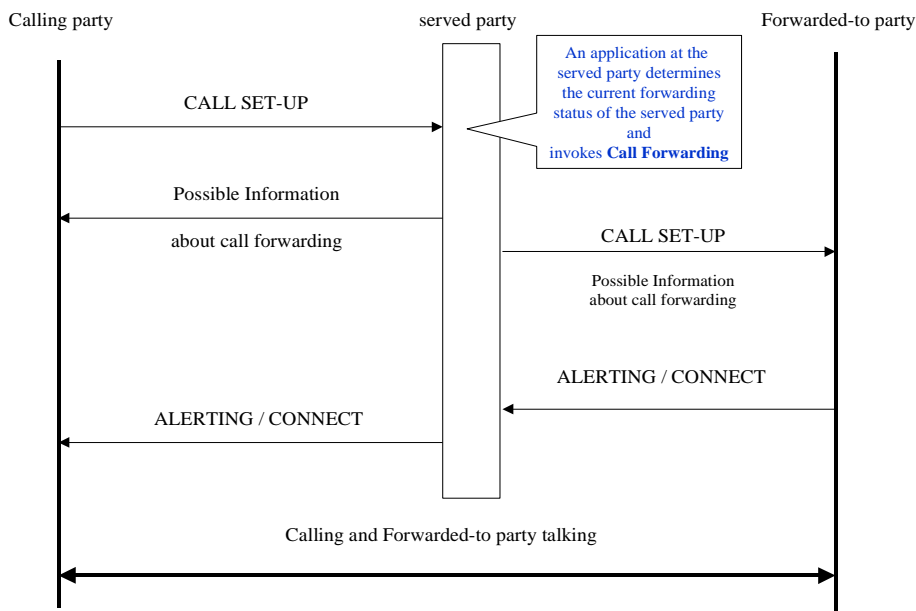

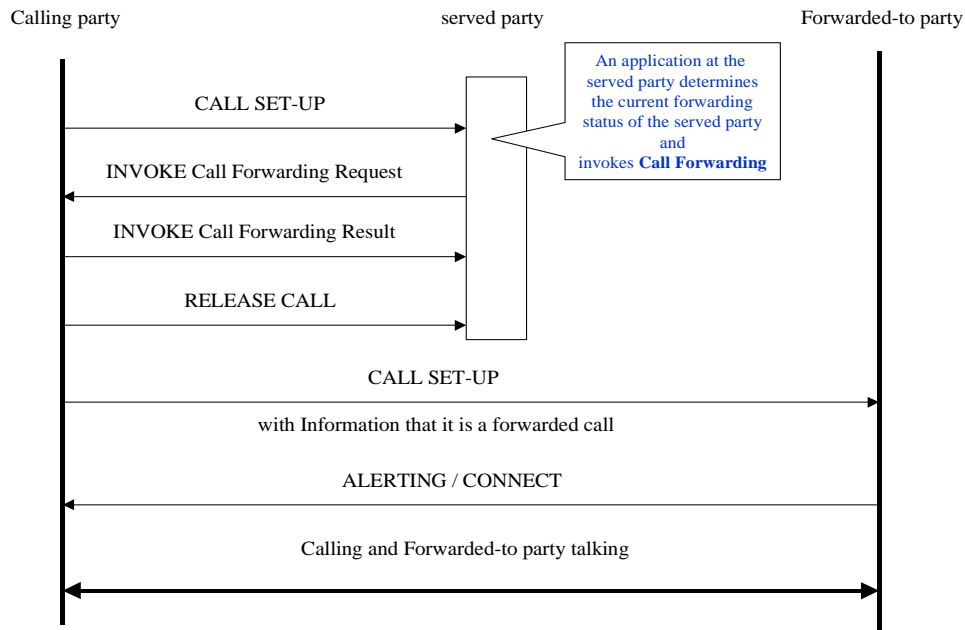Figure 6: Example for call forwarding at the served party

Figure 7: Example for call forwarding with new call set-up between calling and forwarded-to party

D.1.5 Realisation of call forwarding

This chapter shows alternative call forwarding implementation possibilities. In order to have free circulation of terminals and global roaming 3GPP needs to agree on minimum toolkit capabilities. Based on these scenarios different toolkit requirements can be derived.

Roaming would be easier, if all the same toolkits would be available legacy networks, however this assumption does not always apply; specifically it does not apply to ANSI-41. For IM domain to legacy network roaming some toolkit/HLR "conversions" should be studied. They may be rather more complex, but might not need to be standardized.

Note: Scenarios presented below should not be taken as final solutions nor only possible implementation options.

D.1.5.1 Call Forwarding Unconditional

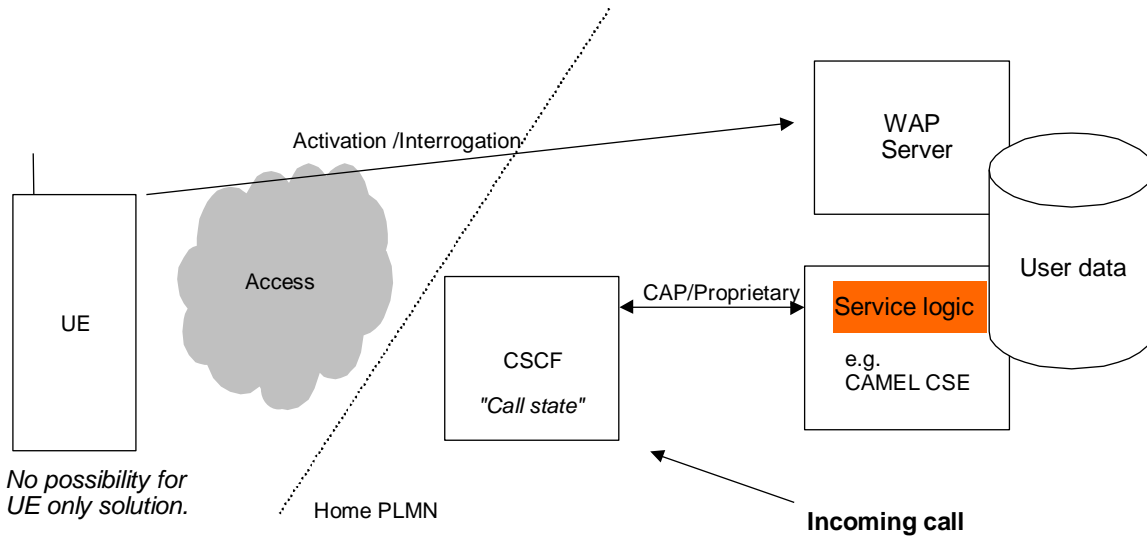Home network diverts calls of selected media types immediately.

Basic scenario

Benefits:
-Allows more complex CF scenarios (time of day,…)

-No roaming impact

Requires:
-Mandatory WAP browser at all phones
- CAMEL at network
or a proprietary solution

WAP Server

User data

Activation /Interrogation

Access

UE

CSCF
"Call state"

CAP/Proprietary

Service logic

e.g.
CAMEL CSE

*No possibility for
UE only solution.*

Home PLMN

Incoming call

D.1.5.2 Call Forwarding No Reply

Network diverts calls of selected media types if there is no reply within selected time limit.
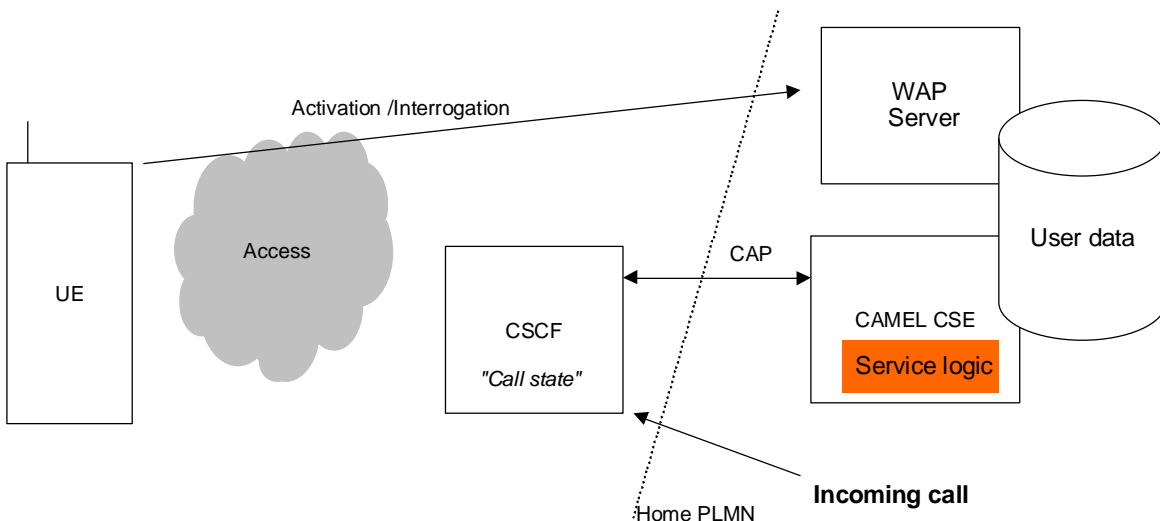
Call Control at VPLMN

Benefits:
-Allows more complex CF scenarios (time of day,…)

-CAMEL can by applied when roaming to e.g. GSM
-Or additional functionality for to transfer CSE logic to/from HLR SS data

Requires:
-Mandatory WAP browser at all phones
-Mandatory CAMEL at network

WAP Server

User data

Activation /Interrogation

Access

UE

CSCF
"Call state"

CAP

CAMEL CSE

Service logic

Home PLMN

Incoming call

Call Control at HPLMN

Benefits:
-Allows more complex CF scenarios (time of day,…)

-CAMEL can by applied when roaming to e.g. GSM
-Or additional functionality for to transfer CSE logic to/from HLR SS data

Requires:
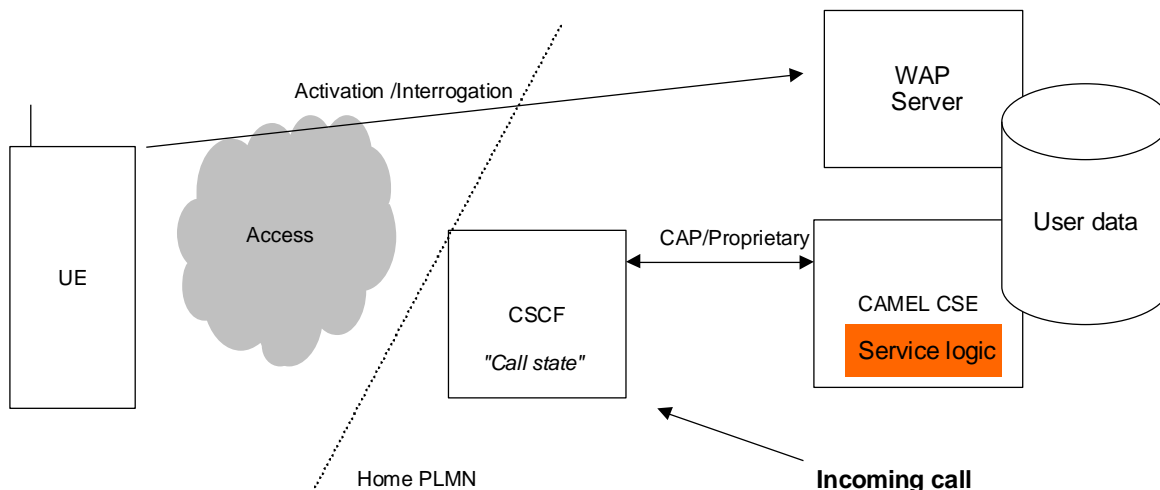-Mandatory WAP browser at all phones
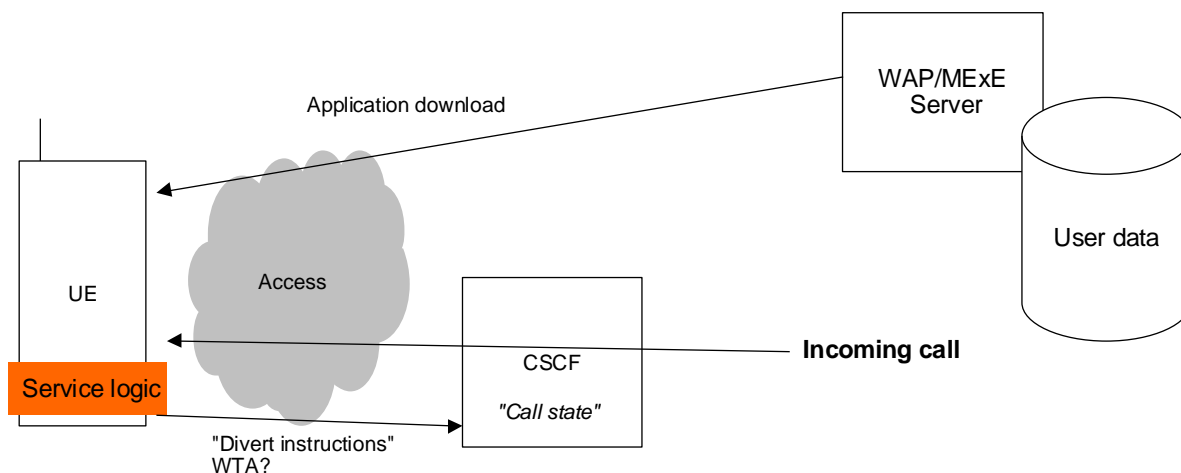-Mandatory CAMEL at network
(also proprietary solution possible)

Terminal based solution

Benefits:
-Allows more complex CF scenarios (time of day,…)

-WAP/MExE can by applied when roaming to e.g. GSM when available (WTA?)
-Or additional functionality for to transfer application logic to/from HLR SS data

Requires:
-Mandatory WAP/MExE at all phones

D.1.5.3   Call Forwarding on Busy

Network diverts calls of selected media types if there is busy signal set by the user. The scenarios are similar to call forwarding on no reply. Only difference is when forwarding is activated, in practise busy situation is detected by the terminal and signalled back to service logic.

Note: There might be no network determined busy as congestion would be unreachable case.

D.1.5.4 Call Forwarding on not reachable

Network diverts calls of selected media types if UE cannot be reached (network congestion, out of coverage, UE switched off etc.). This scenario does not allow UE based solution. The cases when CC is at VPLMN or at HPLMN are similar to call forwarding on no reply case.

History

Document history		
0.0.0	17/11/1999	Skeleton
0.1.0	14/1/2000	Editor's draft input to Rome, Italy R2000 ad-hoc 17-19/1/2000
0.2.0	18/1/2000	Produced during the Rome, Italy R2000 ad-hoc 17-19/1/2000
0.3.0	19/1/2000	Produced during the Rome, Italy R2000 ad-hoc 17-19/1/2000
0.4.0	19/1/2000	Produced during the Rome, Italy R2000 ad-hoc 17-19/1/2000
0.5.0	20/1/2000	Produced during the Rome, Italy R2000 ad-hoc 17-19/1/2000
0.6.0	18/2/2000	Editor's draft input to the Helsinki, Finland R2000 ad-hoc 29/2-2/3/2000
0.7.0	02/3/2000	Produced following the Helsinki, Finland R2000 ad-hoc 29/2-2/3/2000
0.8.0	10/3/2000	Produced during the e-mail discussion period following the Helsinki, Finland R2000 ad-hoc 29/2-2/3/2000
1.0.0	14/3/2000	Version 1.0.0 produced for presentation to SA #7
1.1.0	11/4/2000	Produced during the R2000 ad-hoc at the Beijing, China S1 plenary 10-11/4/2000
1.2.0	4/5/2000	Produced during the Las Vegas, Nevada, USA R2000 ad-hoc 3-5/5/2000
1.3.0	5/5/2000	Produced during the Las Vegas, Nevada, USA R2000 ad-hoc 3-5/5/2000
1.4.0	5/5/2000	Produced during the Las Vegas, Nevada, USA R2000 ad-hoc 3-5/5/2000
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