TSG-SA Working Group 1 (Services) meeting #2 *TSGS1#2(99)~~137~~160*

Edinburgh, Scotland 9th-12th March 1999 Agenda Item: 9.2.2~~0.1~~

CHANGE REQUEST No : ~~A~~003

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

Technical Specification GSM / UMTS: 22.05~~0~~ Version 3.3.0~~2.0~~ ~~.0~~

Submitted to SMG		for approval	X
list plenary meeting or STC here ↑		for information	
without presentation ("non-strategic")		with presentation ("strategic")	

PT SMG CR cover form. Filename: crf26_3.doc

Proposed change affects: SIM [ ] ME [~~X~~] Network [X]

(at least one should be marked with an X)

Work item: ~~GSM evolved network requirements to 3GPP from TTC~~UMTS TS 22.05 Services and Service Capabilities

Source: ~~NEC~~BT

Date: 05 March 1999~~12 Feb., 1999~~

Subject: Addition of new ~~sub section for requirements on emergency call~~text and editorial

Category:

(one category and one release only shall be marked with an X)

F	Correction	
A	Corresponds to a correction in an earlier release	
B	Addition of feature	X
C	Functional modification of feature	
D	Editorial modification	X

Release:

Phase 2	
Release 96	
Release 97	
Release 98	
Release 99	
UMTS	X

Reason for change: ~~The destination of emergency call is different for police station or fire brigade, using two different numbers in Japan (i.e. *110* for police and *119* for fire brigade). It is required to support the capability to identify of emergency call for police or fire brigade in call/connection control~~To enhance the service descriptions and some editorials.

Clauses affected: ~~5.x~~

Other specs affected:

Other releases of same spec		→ List of CRs:
Other core specifications		→ List of CRs:
MS test specifications / TBRs		→ List of CRs:
BSS test specifications		→ List of CRs:
O&M specifications		→ List of CRs:

Other comments:

~~<--------- double-click here for help and instructions on how to create a CR.~~

UMTS 22.05 V3.3~~21~~.1~~0~~ (19998-0316)

Technical Specification

Universal Mobile Telecommunications System (UMTS);
Services and Service Capabilities;
(UMTS 22.05 SMG 1 proposed Vversion 3.3~~12~~.1~~0~~)

~~SA#2 Florida Anticipated Approval - Applied CR A006Version agreed by SMG 1 only~~

UMTS

Universal Mobile
Telecommunications System

ETSI

Reference

DTS/SMG-012205U (kf001io3.PDF)

Keywords

UMTS

ETSI

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Contents

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Foreword

This draft Technical Specification has been produced by the Special Mobile Group (SMG) Technical Committee of the European Telecommunications Standards Institute (ETSI).

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

Version 2.y.z

where:

- x the first digit:
  - 1 presented to SMG for information;
  - 2 presented to SMG for approval;
  - 3 Indicates SMG approved UMTS document.
- y the second digit is incremented for all other types of changes, 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

Pre-UMTS systems have largely standardised the complete sets of bearer services, teleservices and supplementary services which they provide. One major difference between UMTS and pre-UMTS systems is that service capabilities rather than services are standardised for UMTS, allowing service differentiation and system continuity. This Technical Specification (TS) describes how and what kind of services the UMTS user has access to.

2Normative references

References may be made to; References

References may be made to:

a) specific versions of publications (identified by date of publication, edition number, version number, etc.), in which case, subsequent revisions to the referenced document do not apply; or
b) all versions up to and including the identified version (identified by "up to and including" before the version identity); or
c) all versions subsequent to and including the identified version (identified by "onwards" following the version identity); or
d) publications without mention of a specific version, in which case 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] ITU-T recommendation F700 (V xx): "Framework recommendation for audio-visual/ multimedia services".

2.1 Normative references

[21] GSM 02.03: "Digital cellular telecommunications system (Phase 2+); "Teleservices supported by a GSM Public Land Mobile Network (PLMN)".

[3] GSM 02.04;2] GSM 02.04: "Digital cellular telecommunications system (Phase 2+); "General on supplementary services".

[3] GSM 02.42: "Digital cellular telecommunications system (Phase 2+); Network Identity and Timezone (NITZ); Service description; Stage 1".

[4] GSM 02.43: "Digital cellular telecommunications system (Phase 2+); Support of Localised Service Area (SoLSA); Service description; Stage 1".

[5] GSM 02.57: "Digital cellular telecommunications system (Phase 2+); Mobile Station Application Execution Environment (MExE); Service description; Stage 1".

[6] GSM 02.71: "Digital cellular telecommunications system (Phase 2+); Location Services (LCS); Service definition - Stage 1".

[7] GSM 02.78: "Digital cellular telecommunications system (Phase 2+); Customised Applications for Mobile network Enhanced Logic (CAMEL); Service definition - Stage 1".

[8] GSM 02.90: "Digital cellular telecommunications system; Unstructured Supplementary Service Data (USSD) - Stage 1".

[9] GSM 22.01: "Universal Mobile Telecommunications System (UMTS); Service aspects; Service principles".

[10] GSM 22.20: "Universal Mobile Telecommunications System (UMTS); Virtual Home Environment (VHE), Stage 1".

[11] GSM 23.10: "Universal Mobile Telecommunications System (UMTS); UMTS Access Stratum; Services and Functions".

2.2 Informative references

[1] ITU-T recommendation F.700: "Framework recommendation for audio-visual/multimedia services".

[2] GSM 02.01: "Digital cellular telecommunications system (Phase 2+); Principles of telecommunication services supported by a GSM Public Land Mobile Network (PLMN)".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this TS, the following definitions apply:

Basic telecommunication service : this term is used as a common reference to both bearer services and teleservices.

Bearer service : is a type of telecommunication service that provides the capability of transmission of signals between access points.

Call : a logical association between several users (this could be connection oriented or connection less).

Connection : is a communication channel between two or more end-points (e.g. terminal, server etc.).

Mobile termination : the mobile termination is the component of the mobile station which supports functions specific to management of the radio interface (Um).

Multimedia service : Multimedia services are services that handle several types of media. For some services, synchronisation between the media is necessary (e.g. synchronised audio and video). A multimedia service may involve multiple parties, multiple connections, and the addition or deletion of resources and users within a single call.

Nomadic Operating Mode : Mode of operation where the terminal is transportable but being operated while stationary and may in addition require user co-operation (e.g. close to open spaces, antenna setup...).

Quality of Service : the collective effect of service performances which determine the degree of satisfaction of a user of a service. It is characterised by the combined aspects of performance factors applicable to all services, such as;

service operability performance;

- service accessibility performance;
- service retainability performance;
- service integrity performance; and
- other factors specific to each service.

Service feature : Standardised building block used to create services.

Supplementary service : is a service which modifies or supplements a basic telecommunication service. Consequently, it cannot be offered to a user as a standalone service. It must be offered together with or in association with a basic telecommunication service. The same supplementary service may be common to a number of basic telecommunication services.

Teleservice; is a type of telecommunication service that provides the complete capability, including terminal equipment functions, for communication between users according to standardised protocols and transmission capabilities established by agreement between operators.

3.2 Abbreviations

For the purposes of this TS, the following abbreviations apply;

BER	Bit Error Rate
B-ISDN	Broadband ISDN
CAMEL	Customised Application for Mobile network Enhanced Logic
DTMF	Dual Tone Multiple Frequency
TR	Technical Report
TS	Technical Specification
ETSI	European Telecommunications Standards Institute
FAX	Facsimile
GSM	Global System for Mobile Communications
HE	Home Environment
IMUN	International Mobile User Number
IN	Intelligent Network
ISDN	Integrated Services Digital Network
ISO	International Organisation for Standardisation
ITU	International Telecommunication Union
LCS	Location Services
MExE	Mobile station Execution Environment

MMI	Man Machine Interface
MO	Mobile Origination
MS	Mobile Station
MT	Mobile Termination
O&M	Operations and Maintenance
PBX	Private Branch eXchange
PC	Personal Computer
PCMCIA	Personal Computer Memory Card International Association
PIN	Personal Identity Number
PNP	Private Numbering Plan
POTS	Plain Old Telephony Service
QoS	Quality of Service
USIM	User Service Identity Module
SMS	Short Message Service
SAT	SIM Application Toolkit
SN	Serving Network
SoLSA	Support of Localised Service Area
UMTS	Universal Mobile Telecommunications System

4 Framework for the description of telecommunication services and applications

4.1 General

Telecommunication services supported by UMTS are the communication capabilities made available to users by network operators and service providershome environment andserving network. A UMTS network provides, in co-operation with other networks, a set of network capabilities which are defined by standardised protocols and functions and enable telecommunication services to be offered to users.

A service provision by a service provider/network operatorHE/SN to a UMTS user may cover the whole or only part of the means required to fully support the service. The operational and commercial features associated with the provision of the service are included in the service concept.

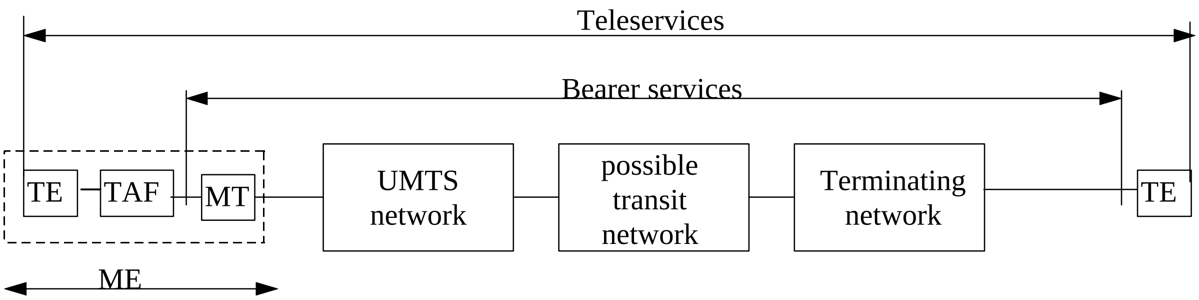
The service classification and description which follow are independent of different possible arrangements for the ownership and provision to the user of the means required to support a service.

4.2 Basic telecommunication services

Basic telecommunication services are divided in two broad categories;

- bearer services, which are telecommunication services providing the capability of transmission of signals between access points;
- teleservices, which are telecommunication services providing the complete capability, including terminal equipment functions, for communication between users according to protocols established by agreement between network operators.

Figure 1 illustrates these definitions.



ME: Mobile Station
MT: Mobile Termination
TE: Terminal Equipment
TAF: Teminal Adaption Function

NOTE 1: In order to limit the complexity of the figure, only one transit network is shown.
NOTE 2: The terminating network type may include a UMTS network, either the originating one or another one.
NOTE 3: The bearer service terminates in the mobile station.

Figure 1; Basic telecommunication services supported by a UMTS network

4.2.1 Bearer services

The characterisation of a bearer service is made by using a set of attributes. A bearer service attribute is a specific characteristic that distinguishes it from other bearer services. Particular values are assigned to each attribute when a given bearer service is described and defined.

The attributes define the service characteristics as they apply at a given reference point where the user accesses the bearer service. The description of a bearer service by the method of attributes is composed of technical attributes.

A list of definitions of attributes and values used for bearer services is contained in clause 5.

The bearer services are negotiable and can be used flexibly by applications.

4.2.2 Teleservices

Clause 6 defines both standardised and non-standardised teleservices. Some teleservices are standardised because that interworking with other systems have been recognised as a requirement. Other teleservices shall not be standardised. A decoupling between lower layer (i.e. bearer attributes) and higher layer capabilities will be necessary for the development of teleservices.

4.3 Supplementary services

A supplementary service modifies or supplements a basic telecommunication service. Consequently, it cannot be offered to a user as a stand alone service. It must be offered together or in association with a basic telecommunication service. The same supplementary service may be applicable to a number of basic telecommunication services.

Two methods are used for the characterisation of supplementary services;

- The first method is used for the description of existing standardised supplementary services. These services are specified through the detailing of each of the operations involved in service provision and service usage (the provision/withdrawal, registration/erasure, activation/deactivation, invocation and interrogation operations). Clause 7 lists these services.
- The second method enables the provision of service provider/network operatorHE/SN specific supplementary services. To make this possible, standardised building blocks referred to as service features are specified in clause 8. The combination and parametrisation of these service features allow the creation of supplementary services.

UMTS shall be able to handle multiple supplementary services within a call. Interactions shall be handled when several supplementary services are activated in the same call.. When multiple supplementary services can be activated concurrently, some prioritisation of the services will be necessary. Certain services may override or deactivate other services.

Interactions between operator specific supplementary services are not defined.

The following issues need consideration when interactions between services occur;

- Different phases of a call.
- A service spanning on more than one network.

- Service interactions that may occur between services offered to a single user, as well as between services offered to different interacting users.

NOTE: The methods defined for characterisation of services are description methods. They do not imply or restrict different implementations.

4.4features Service features

UMTS service features are based on functionality and mechanisms such as provided by SAT, MExE, IN and CAMEL. These toolkits are the basic building blocks for the VHE. These features can be used both by standardised and non-standardised services.

through the UMTS Application Programming Interface. The UMTS services and applications get access to UMTS service capabilities 5(bearers) for transport of user data through the UMTS adaptation layer. This lowest layer of the VHE is responsible for the selection of appropriate service capabilities according to the requirements of services and applications.

High level service features requirements :

. support of wide range of user applications,
. support of rapid application/service development,
. support of easy deployment of new services,
. scalability.

5 Bearer Services

5.1 Definition of bearer services

Bearer services provide the capability for information transfer between access points and involve only low layer functions. These functions are sometimes referred as low layer capabilities (in reference to OSI layers). The user may choose any set of high layer protocols for his communication and the UMTS network does not ascertain compatibility at these layers between users.

5.2 Description of bearer services

Bearer services are characterised from a static point of view by a set of low layer attributes. This set has been chosen so that a bearer service can be entirely defined by giving a value to each attribute of the set. In particular, the set and the associated allowed values enable characterisation of future (not yet used or foreseen) transfer needs.
Giving one of the possible values to each attribute defines a possible bearer service. However, any combination is neither meaningful nor necessarily supported by the UMTS system. This section defines the attributes and their possible values. The authorised combinations are specified in the following sections.
The parameters of the set are grouped into two categories;

- Information transfer attributes, which characterise the ~~network transfer capabilities~~QoS required for transferring user information between two or more access points.
- Information quality attributes, which characterise ~~the quality of the user information transferred~~the QoS obtained between two or more access points.

Most of the attributes presented further down may be attributed several values when the bearer service required by an application involves more than one traffic type (connection/connectionless) or more than one connection.
It shall be possible to negotiate/re-negotiate all of the attributes presented in this clause at call set-up/ during the call (mobile or network initiated).

5.2.1 Information transfer attributes

Connection mode attribute

The two possible values for this attribute are connection oriented and connectionless. In a connection oriented mode, information is delivered to the destination entity in the same order as it was provided by the source entity, but an establishment/release phase is required at the beginning and the end of the information transfer. In a connectionless mode, information can directly be transferred, but with no guaranty of ordered delivery.

Traffic type attribute

The four possible values for this attribute are constant bit rate, variable bit rate, available bit rate and unspecified bit rate.

Symmetry attribute

The three possible values for this attribute are unidirectional, bi-directional symmetric and bi-directional asymmetric.

Communication configuration attribute
This attribute indicate the spatial arrangement for transferring information between the implicated access points. The possible values are point-to-point, and point-to-multipoint. When the value of the attribute is point-to-multipoint, it shall be further characterised as multicast or broadcast. The addresses of the source entity and the destination entities should also be provided. One multipoint address should be reserved for broadcasting.

Information transfer rate attributes
Information transfer rate is the amount of information transmitted per unit of time from a source access point to destination access point(s).
The three attributes used to characterise the information transfer rate are the peak bit rate, the minimum bit rate and the mean bit rate. . The possible values for these three attributes are not a limited set, but a continuous range of values. More parameters may certainly be needed, such as the sustainable bit rate or the occupancy (FFS).

5.2.2 Information quality attributes

Information quality attributes characterise the bit integrity and delay requirements of the applications.
Other parameters may be needed.

Maximum transfer delay attribute
This attribute sets the maximum transfer delay of the information. The two reference points for the maximum transfer delay are the Iu interface and the point located between the mobile termination and the terminal adaptation function. The possible values for this attribute are not a limited set, but a continuous range of values.

Delay variation attribute
This attribute sets the variation in the received information. This attribute is important for real-time services, e.g. video conference, where a value approaching 0 would typically be requested. The possible values for this attribute are not a limited set, but a continuous range of values.

Bit error ratio attribute
The ratio between incorrect and total transferred information bits. The possible values for this attribute are not a limited set, but a continuous range of values.

Error characteristics attribute
This attribute characterises the arrivals of errors. The two possible values are uniform and bursty.

5.3 ~~Supported~~ Required bit rates

It shall be possible for one application to specify its traffic requirements to the network by requesting a bearer service with any value for the connection mode, traffic type, symmetry and information transfer rate attributes. It shall be possible for the network to satisfy these requirements without wasting resources on the radio and network interfaces due to granularity limitations in bit rates.
It shall be possible for one mobile termination to have several active bearer services simultaneously, each of which could be connection oriented or connectionless.
The only limiting factor for satisfying application requirements shall be the cumulative bit rate per mobile termination at a given instant (i.e. when summing the bit rates of one mobile termination's simultaneous connection oriented and connectionless traffic, irrespective of the traffic being real time or non real time) in each radio environment :

- At least 144 kbits/s in satellite radio environment (Note 1).
- At least 144 kbits/s in rural outdoor radio environment.
- At least 384 kbits/s in urban/suburban outdoor radio environments.
- At least 2048 kbits/s in indoor/low range outdoor radio environment.

NOTE 1 : This Peak Bit Rate may only be achieved in a nomadic operating mode.

5.4 ~~Supported~~ Required QoS

It shall be possible for one application to specify its QoS requirements to the network by requesting a bearer service with any value for the maximum transfer delay, delay variation, bit error rate and error characteristic attributes.
The following table indicates the range of values that shall be supported by UMTS for the QoS attributes. These requirements are valid for both connection and connectionless traffic. It shall be possible for the network to satisfy these requirements without wasting resources on the radio and network interfaces due to granularity limitations in QoS.

	Real Time (Constant Delay)	Non Real Time (Variable Delay)
Operating environment	**BER/Max Transfer Delay**	**BER/Max Transfer Delay**
Satellite (Terminal relative speed to ground up to 1000 km/h for plane)	Max Transfer Delay less than 400 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 1200 ms or more (Note 2) BER = 10-5 to 10-8
Rural outdoor (Terminal relative speed to ground up to 500 km/h) (Note 3)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 150 ms or more (Note 2) BER = 10-5 to 10-8
Urban/ Suburban outdoor (Terminal relative speed to ground up to 120 km/h)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 150 ms or more (Note 2) BER = 10-5 to 10-8
Indoor/ Low range outdoor (Terminal relative speed to ground up to 10 km/h)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 150 ms or more (Note 2) BER = 10-5 to 10-8
NOTE 1; There is likely to be a compromise between BER and delay. NOTE 2; The Max Transfer Delay should be here regarded as the target value for 95% of the data. NOTE 3; The value of 500 km/h as the maximum speed to be supported in the rural outdoor environment was selected in order to provide service on high speed vehicles (e.g. trains). This is not meant to be the typical value for this environment (250 km/h is more typical).		

The attributes of the services that will be provided within these environments are:

- **Voice** (one to one and one to many, voicemail, dictation)
  - Very low delay requirement, low delay variance requirement, real time service so better to have errors rather than re-transmit. However error rate must also be low, symmetrical traffic profile
- **Fax** (fax, also internet & LAN access at low speed)
  - Low delay requirement, but less so than for voice . Errors could be corrected as long as the user pervices service within acceptable delay bounds), symmetrical traffic profile
- **Interactive** (high quality audio conference, video conference, remote expert, multimedia call centre,
  - collaborative working, telepresence (e.g. telemedicine). Some applications with a lower volume throughput may also fall into this category as a result of delay requirements, e.g. games, e-commerce (when time critical), booking services (again could be time critical). Synchronisation may be requireed between streams, but often this is done by application layer protocols, e.g. H.263 suite
  - Low delay requirement, delay variance is important and its impact depends on the size of buffer used on the receiving system. A larger buffer can tolerate greater variance in delay, but will provide greater overall delay to the receiver. Symmetrical traffic profile.

Messaging (SMS style messaging, alerting, email without attachments, diary & schedule updates,. Dispatch, tracking, information search, telemetry)
  - Delay tolerant as these are store and forward services. Asymmetric. Small data volume. Can tolerate higher error rates as long as these can be corrected by higher layer protocols.
- **Medium multimedia** (Internet / Intranet/ remote LAN access, email with attachments, groupware, mobile office, applet download)
  - Delay tolerant, subject to user perceiving acceptable QoS. Tolerant to delay as long as higher layers can correct within QoS tolerances. Highly asymmetric, 90% “downlink”.
- **High multimedia** (High speed internet/intranet/LAN access, streamed audio, video, software download,)

- Delay tolerant, subject to user perceiving acceptable QoS. Tolerant to delay as long as higher layers can correct within QoS tolerances. Highly asymmetric, 90% “downlink”. Much larger data volumes than medium multimedia, higher quality services.
- Need to quantify the QoS parameters for these services. Typical packet sizes. UMTS should allow transfer of variable size packets, including all TCP/IP protocols.

5.5 Supported topologies

It shall be possible for an application to specify its traffic topology requirements to the network by requesting a bearer service with any value for the communication configuration attribute. However, some combinations with the symmetry attribute are not authorised. The supported configurations are :

1) Point-to-Point
   - Uni-Directional
   - Bi-Directional
     - Symmetric
     - Asymmetric
2) Uni-Directional Point-to-Multipoint
   - Multicast
   - Broadcast

A multicast topology is one in which sink parties are specified before the connection is established, or by subsequent operations to add or remove parties from the connection. The source of the connection will always be aware of all parties to which the connection travels.

A broadcast topology is one in which the sink parties are not always known to the source. The connection to individual sink parties is not under the control of the source, but is by request of each sink party.

In the case of a mobile termination with several active bearer services simultaneously, it shall be possible for each bearer service to have independent topologies and source/sink parties.

5.6 Radio Interface optimisation

The following requirements shall lead the radio interface optimisation process;

- support of high bit rate (around the Peak Bit Rate), bursty, asymmetric, non-real time bearer capabilities;
- support of high bit rate (around the Peak Bit Rate), bursty, asymmetric, real time bearer capabilities;
- the ability to extend or reduce bandwidth associated to a bearer capability in order to adapt to bit rate or radio condition variations, to add or drop service components.

However, the services provided by GSM (speech in particular) shall be supported in a spectrally efficient manner (at least as efficiently as in GSM) for the same quality of service.

In order to allow the support of flexible, bandwidth on demand services, bearer services should be provided with the finest possible granularity that can be efficiently supported.

6 Teleservices

6.1 Definition of teleservices

Teleservices provide the full capacity for communicationbilities for communications by means of terminal equipment and, network functions and possibly functions provided by dedicated centres.

6.2 Description of teleservices

The basic reference in UMTS for the description of teleservices is the ITU-T F700 recommendation. F700 provides a generic, network independent, description of multimedia services. The methodology used covers both monomedia and multimedia services, the monomedia services being a particular type of multimedia services. Multimedia services are classified into categories with similar functional characteristics. The six categories are multimedia conference services, multimedia conversational services, multimedia distribution services ,multimedia retrieval services, multimedia messaging services and multimedia collection services.

The rest of clause 6 describes the teleservices and options that will be provided by UMTS networks.

A teleservice can be viewed as set of upper layer capabilities utilising the lower layer capabilities described by the set of attributes in clause 5.

Multimedia teleservices support the transfer (and in some case retrieval, messaging, distribution) of several types of information (service components). For this reason, there are service attributes (relating to all the components of a teleservice) and service component attributes (relating to only one service component).

6.3 Support of teleservices in UMTS networks

The realisation of teleservices requires the association of terminal and network capabilities. In the terminals and in the network, both upper layer capabilities and lower layer capabilities are necessary. The term upper layer capabilities is used because it relates to the OSI upper layers. Decoupling between upper layers and lower layers (transfer) is required. Even if this de-coupling may impact radio interface optimisation, it is nevertheless the only way of designing a system that is not outdated;

- Each time the information rate associated with an already supported teleservice is decreased by more efficient source coding techniques.
- Each time a new service is introduced that requires transfer capabilities not used by currently available teleservices.

Taking the example of two application that exchange information through a teleservice, the upper layer capabilities can be located in various places;

- In the two terminals if the two applications are connected to a UMTS network.

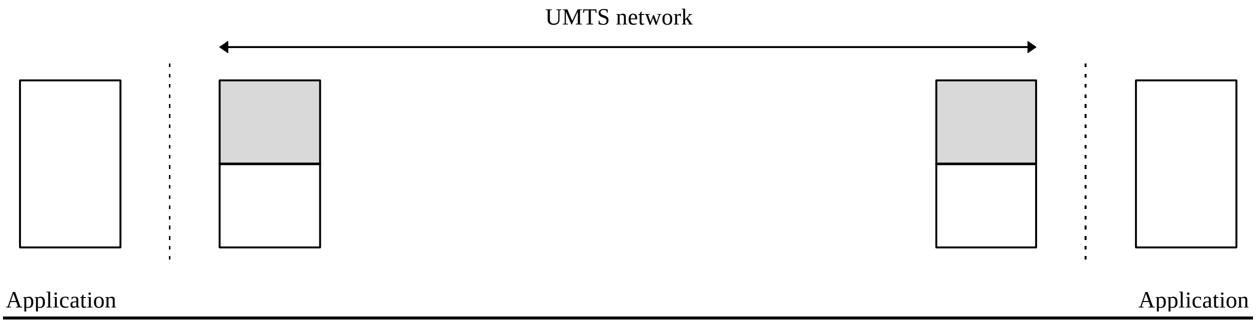


Figure 2; UMTS teleservice

In the terminal of the application connected to a UMTS network and in the upper layer interworking unit that is at the border of the UMTS network and the target network if one application is connected to a UMTS network and the other one is connected to another type of system. The upper layer interworking unit makes the adaptation between the UMTS network and the target network at a service level.

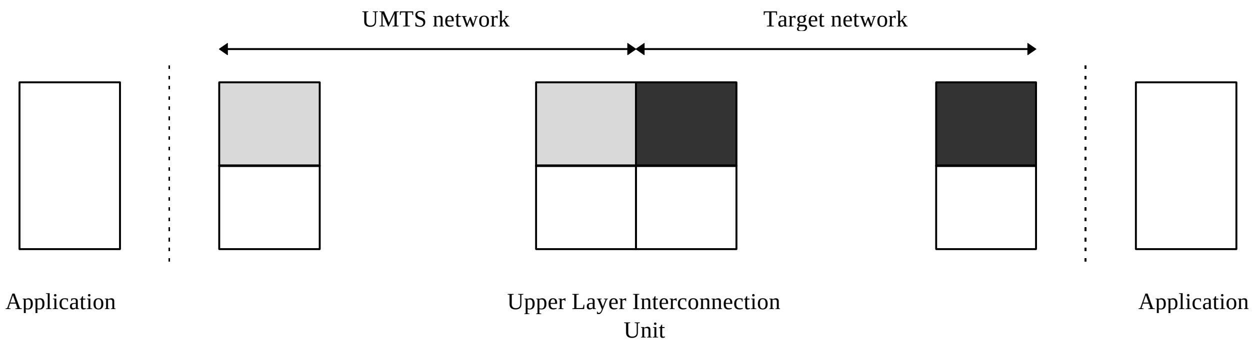


Figure 3; Teleservice with upper layer interworking

In the terminal of the application connected to a UMTS network and in the terminal of the application connected to a target network if one application is connected to a UMTS network and the other one is connected to another type of system, but only lower layer interconnecting unit is used at the border of the two networks. In this case, the interconnecting unit makes the adaptation between the UMTS network and the target network at the transmission level.

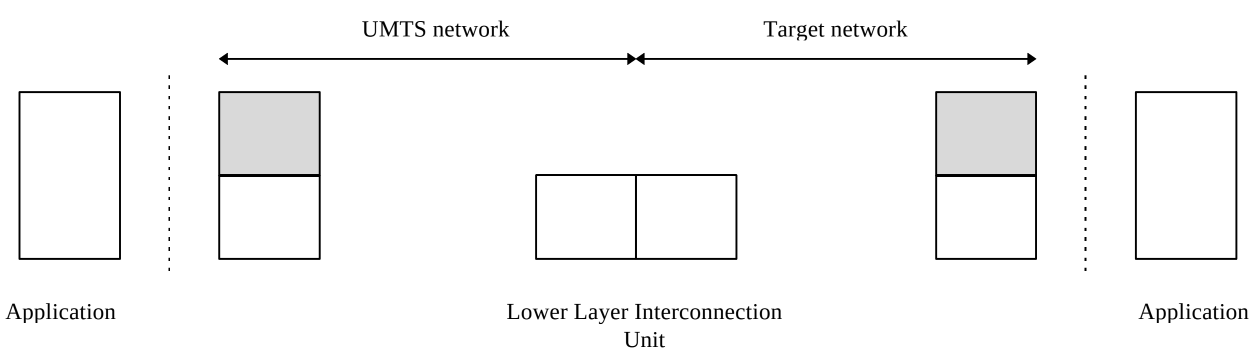


Figure 4; Teleservice with lower layer interworking

6.4 Existing Teleservices supported by UMTS networks

The subset of standardised teleservices shall be supported by UMTS for interworking with teleservices provided on other networks. The means to support the following set of teleservices will be standardised;

- Speech;
- Emergency call;
- Short message service;
- Facsimile.

6.4.1 6.4.1 Speech

The speech service as defined in international standards should be supported by UMTS. The international reference for the speech is ITU E.105 recommendation. UMTS networks should contain interworking units which allow calls to be received from or destined to users of existing networks like PSTN, ISDN, GSM. This will include interworking units for generation of DTMF or other tones (the entire DTMF tone set would at minimum be available) and detection of DTMF tones.

Top level essential requirement is for wireline quality speech in home, office and public building. It is also highly desirable to provide wireline quality speech whilst outdoors or on the move.

A default speech codec is required for UMTS (refer to 22.25 for QoS)shall be specified to provide speech service across the UTRAN. The selected speech codec shall be capable of operating with minimum discernible loss of speech on handover between the GSM access network and UTRAN.

6.4.2 Emergency Call

This service will use components of Speech. There are however compared to Telephony reduced authentication requirements and a requirement for specific routing. Additionally Emergency Calls may have higher priority than normal calls, etc.. The reference for the emergency call service is GSM 02.03.

6.4.3 Short Message Service - Point to Point (SMS-PP)

The short message service point to point as specified in the 02.03 shall be supported in UMTSGSM 02.03 shall be supported in UMTS. A short message service shall be provided seamlessly (as far as the user or the users terminal equipment is concerned) across ~~the~~ the UMTS ~~and GSM~~ access networks. Additional features are planned for SMS in GSM Release 99.

6.4.4 Short Message Service - Cell Broadcast (SMS-CB)

A short message service cell broadcast shall be provided seamlessly (as far as the user or the users terminal equipment is concerned) across the UMTS and GSM network.

4.4 Facsimile

Facsimile group 3, for interworking with fax in the fixed network, shall be supported. The Fax service is described in ITU F180 and F180 bis recommendations.

6.5 Internet Access

Internet is seen as the most important source of data traffic in UMTSUMTS shall provide means to interwork with external data networks. This interworking shall satisfy, within the constraints introduced by the mobile radio environment, the QoS requirements of the interworked-with network. For UMTS the Internet is seen as the most important interworked-with network, therefore the specification of an optimised access to Internet shall be part of the UMTS standard. The most important benefits achieved by the definition of Internet Access would be:

- Optimised transmission of IP traffic over the UMTS radio interface to minimise the amount of information transmitted and minimise delay..
- Optimised usage of encryption protocols/algorithms over the UMTS radio interface.
- Inter-operation of QoS mechanisms used in both, UMTS and in Internet.
- Provision of IP addresses for the UMTS users using Internet applications and accessing Internet.

For the purposes of optimised access to Internet one or more of the UMTS generic bearers will be used. On top of the bearer a UMTS protocol profile will be defined. This profile would be based on the work done by IETF or other relevant fora, and will consist of a recommended set of parameters and standardised protocols providing similar services than the Internet ones but optimised for wireless access. In the case of Internet traffic it would be possible for the user to select the encryption to be used (e.g. no encryption, end-to-end encryption, encryption over UMTS radio, etc.). The QoS mechanisms defined for UMTS packet access mode shall be harmonised with those defined for Internet (e.g. RSVP, RTP). UMTS users shall be provided with IP addresses for enabling the usageDifferentiated Services).
of IP applications.

For multimedia applications, simultaneous delivery of more than one media is required.

There is a requirement to adapt data speed during a multimedia session – ensuring that the session is maintained despite possible change in quality.

7 Supplementary Services

Supplementary services are used to complement and personalise the usage of basic telecommunication services (bearer services and teleservices). The capabilities standardised in UMTS shall enable all the supplementary services specified in GSM 02.04 and the 02.8x set and GPRS *** to be provided.

The GPRS Stage 1 identifies the following GSM phase 2 supp services for support on GPRS

Table 9: GSM Phase 2 Supplementary Services

Supplementary Services		Applicability
CLIP	Calling Line Identification Presentation	NA (note 1)
CLIR	Calling Line Identification Restriction	NA (note 1)
CoLP	Connected Line Identification Presentation	NA (note 1)
CoLR	Connected Line Identification Restriction	NA (note 1)
CFU	Call Forwarding Unconditional	A (note 5)
CFB	Call Forwarding on Mobile Subscriber Busy	NA (note 2)
CFNRy	Call Forwarding on No Reply	NA (note 2)
CFNRc	Call Forwarding on Mobile Subscriber Not Reachable	A (note 5)
CW	Call Wait	NA (note 3)
HOLD	Call Hold	NA (note 3)
MPTY	Multi-Party Service	NA (note 4)
CUG	Closed User Group	A
AoCI	Advice of Charge - Information	A
AoCC	Advice of Charge - Charging	A
BAOC	Barring of All Outgoing Calls	NA (note 6)
BOIC	Barring of Outgoing International Calls	NA (note 6)
BOIC-exHC	Barring of Outgoing International Calls except those directed to the HOME PLMN Country	NA (note 6)
BAIC	Barring of All Incoming Calls	NA (note 6)
BIC-Roam	Barring of Incoming Calls when Roaming Outside the HOME PLMN Country	NA (note 6)

NOTE 1: These are not applicable because it is an integral feature of packet data protocols that a PDU arrives at the terminating DTE with a Source and Destination address included. This is similar to SMS.
NOTE 2: These supplementary services may have no meaning in a packet orientated network. For example, the "busy" condition (CFB), as defined in a circuit-switched network, is not possible in a packet data network. Instead, packets experience increased throughput delay.
NOTE 3: The interworking protocols of an external data network make these supplementary services unusable. For example, X.25 networks will time-out, assume a PDU is lost, and either retry or abandon transmission altogether if a significant delay is experienced.
NOTE 4: The PTM-G services cover the requirements of Multiparty applications.
NOTE 5: Call Forwarding may not be applicable when interworking with certain external data networks.
NOTE 6: These are part of the message screening function (see subclause 5.4.11). The concept of call barring (in the sense of circuit switched services) may not be applicable to GPRS with all external data network protocols.

Key: **NA:** Not Applicable **A:** Applicable

8 Service features

Service features are building blocks which can be used to create/control/delete services. The functionality offered by a service feature may depend upon the underlying service capability used to realise the service feature e.g. CAMEL, MExE etc.. Service features may be used to offer the user some control over a service such as the ability to modify a service, subscribe or unsubscribe to a service.

Service features are associated with call/session control, bearer control, mobility management. The term calls is used to encompass not only circuit-switched (e.g. voice) calls, but also virtual-circuit sessions set-up to handle packet data traffic.

As a minimum t~~T~~he following service features are required;

- security/privacy;
- access control;
- address translation;
- call/session/bearer control;
- location;
- messaging;
- service control;
- user interaction.

8.1 Security/Privacy features

- presentation of or restriction of information associated with a party involved in a call or a session (e.g. calling line ID, calling name, location...);
- encryption of user data and signalling;

(From GPRS stage 1) The use of radio communications for transmission to/from subscribers in mobile networks makes them particularly sensitive to:

1) misuse of their resources by unauthorized persons using manipulated MSs;
2) eavesdropping on the information being exchanged on the radio path.

Therefore, to protect the system in the two cases mentioned above, the following security features are provided for GPRS:

- MS authentication; i.e., the confirmation by the land-based part of the system that the subscriber identity, transferred by the MS within the identification procedure on the radio path, is the one claimed. The purpose of this authentication is to protect the network against unauthorized use. It also enables the protection of GPRS subscribers by denying intruders the ability to impersonate authorized users;
- access control; i.e., the network can support restrictions on access by or to different GPRS subscribers, such as restrictions by location, screening lists, and so on;
- user identity confidentiality; i.e., the property that the user identity on the radio link is not made available or disclosed to unauthorized individuals, entities or processes. The purpose is to provide privacy of identities of the subscribers who are using GPRS radio resources. It allows for the improvement of other security features, e.g., user information confidentiality, and also provides for the protection against tracing the location of a mobile subscriber by listening to the signalling exchanges on the radio path;
- user information confidentiality; i.e., the property that the user information is not made available or disclosed to unauthorized individuals, entities or processes. The purpose is to provide for confidentiality of user data, i.e., protection of the message part pertaining to layers 3 and above, that passes over the radio path.

Both user identity and user data shall be protected as shown in table 6:

Table 6: Protection of user identity and user data

Service	User Identity Protection	User Data Protection
PTP	Yes	Yes
PTM-Multicast (receiver)	Yes [a)]	No[b)]
PTM-Group Call	Yes	Yes

a) The individual identities of the group members that actually receive the PTM-M traffic, are not transferred on the radio path and furthermore are also not known to the network. This is an important aspect for those applications where it is imperative that the location of the user cannot under any circumstances be traced. However, the group identity and the identity of the service requester are sent unciphered on the radio path.

b) This does not preclude end-to-end ciphering of user data by the PTM-M application, this however, is outside the scope of this specification.

Security mechanisms available for existing teleservices and bearer services should be used if possible.

An optional requirement is that an MS can anonymously initiate a mobile originated, PTP communication to a specific subscriber or server that is registered within the PLMN. This necessitates that all charges shall be made to the called party. For the access to the network the MS shall not send its IMSI or IMEI thus guaranteeing a high level of anonymity. However, in the case of fraud or misuse of the service, the MS shall transfer its IMEI and/or IMSI upon request by the operator. Authentication and ciphering procedures are not required. Such procedures may reside inside or outside the GSM network.

NOTE: An example for such an anonymous service is a toll road system whereby a user can pay the road-toll anonymously using a pre-paid card instead of a normal SIM card. The road-toll application server receives and is charged for all messages of the anonymous service. However, the server has its own means to charge the user (e.g. using electronic money on the anonymous pre-paid card).

5.4.11 Message Screening

The message screening function is concerned with filtering out unauthorized or unwanted messages. Message screening may be used to restrict the types of message or the volume of data which may be transferred across the GPRS network to/from an individual subscriber.

5.4.11.1 Network controlled screening

The PLMN administration and/or the GPRS service provider shall be able to provide basic screening functionality (e.g. firewall) to reduce the risk of fraud and misuse, to ensure the integrity of the network and to protect subscribers.

5.4.11.2 Subscription controlled screening

The subscriber is able to negotiate through the subscription, specific message screening requirements within the limitations of the network (HPLMN or VPLMN) controlled screening. The subscriber shall be able to limit the amount of chargeable data, the source, the destination and the types of messages sent or received. The subscription controlled screening is applicable also when roaming.

It shall be possible for the subscriber to define screening criteria for each interworking profile and for the non-interworking case.

5.4.11.3 User controlled screening

The user is able to individually control message screening within the limitations of the network (HPLMN or VPLMN) and the subscription controlled screening. The user shall be able to limit the amount of chargeable data, the source, the destination and the types of messages sent or received. The user controlled screening is applicable also when roaming.

- It shall be possible for the user to define screening criteria for each interworking profile and for the non-interworking case.

8.2 Access Control features

The access control features are defined to provide access to the UMTS network to the UMTS users over the serving network's air interface. These features include;

- user registration;
- user de-registration;
- authentication (any actor in the role model shall be able to authenticate with any other actor with whom she has direct communication)muthual authentication.

8.3 Address Translation Features

e parties may be addressable via different means, they should be reachable independent of the medium. To support this requirements a new network functionality which can map the address, name (alpha string) / number (digits) and service type onto a service provider for call routing.

This address translation functionalityeature shall allow UMTS to offer the wide range of addressing options including;

- E.164 Numbering (e.g. GSM MS-ISDN);
- ASEA Numbering (ATM);
- IP v6 Numbering;
- X.25 Numbering;
- Internet symbolic naming.

The content of this clause will be updated as the result of 22.75 conclusions.

8.4 Call/Session/Bearer Control Features

These features will be used to establish, handle and terminate calls. The following service features shall be supported;

- call/session set-up (point to point, point to multi-point, multi-point to multi-point);
- add/delete a party from a call/session;
- call/session termination;
- call/session establishment e.g. answering of calls;
- monitoring of call/session states and events;
- modification of the bearer service attributes.
- capability at initial call set-up to modify or reject the called party address;
- capability for an incoming call to modify or reject the called party address both at early and late stage of the call;
- capability to suspend and resume a call;
- capability to re-route a call;
- capability to be notified when a specified terminal is free or is ready to accept the call.

8.5 Location Features

Location capabilitifeatures shall also be supported, to allow new and innovative location based services to be developed;

- to identify and report in a standard format (e.g. geographical co-ordinates) the current location of the user's terminal.

The precision of the location shall be network design dependent, i.e. an operator choice. This precision may vary from one part of a network to another. It may be chosen to be as low as hundreds of meters in some place and as accurate as 5 meters in other place. It is required that a minimum precision of around 50 meters can be achieved in all types of terrestrial radio environment. Technical issues may constrain the precision to be mobile state dependent as well (mobile idle / mobile in communication). Several design optional features (e.g. size of the cell, adaptive antenna technique, path loss estimation technique...) shall allow the network operator to reach cost effectively the target precision.

Because there may be very different uses of the location information;

- It shall be possible to make the information available to the user, network operator, service providerHE/SN and value added service providers. The user shall be able to restrict access to the location information (permanently or on a per call basis). The restriction can be overridden by the network operator when appropriate (e.g. emergency calls).
- It shall be possible to set the delay to get the location information (the situation is quite different whether the information is needed for call routing or if it is needed by a user application).
- It shall be possible to select the frequency of the location information update.

If the terminal is switched off, then the last known position and time/date shall be available. The time of last known location shall be recorded and be made available in universal time.

- to identify and report when the user’s terminal enters or leaves a specified geographic area.
- It shall be possible to specify the area as a circular zone (centre and radius) to a resolution that will be limited by the accuracy capability of the part of the serving network where the user is registered.

8.6 Messaging features

UMTS shall support the use of exisitng messaging systems (email, groupware etc).

Messages are a block of data that may range from a few bytes to megabytes. Message delivery may involve store and forward of messages in transit. To be able to exchange and to control the exchange of messages between user the following service features shall be supported;

- capability to send messages;
- capability to receive messages;
- capability to request confirmation of receipt;
- capability to modify the content as well as the recipient of message;
- capability to reject a outgoing and/or incoming message;
- capability to re-route a message.

8.7 Service control features

To allow the support of service provider/network operatorHE/SN specific services the following service features shall be supported;

- capability to download service software to network nodes;
- capability to download service software to terminals;
- capability to download service software to the USIM;
- capability to negotiate of supported capabilities between USIM, terminals, serving network and service provider.HE and SN;
- capability to negociate bearer services and service capabilities

8.8 User Interaction Features

To allow the support of service provider/network operator specific user interfaces the following service features shall be supported;HE/SN specific user interfaces, databases containing user profiles shall be provided. This user profile functionality shall provide the following interaction features :

- capability to indicate information to the user;
- capability to collect user information;
- capability to activate and deactivate a special user profile;
- capability to change the user profile.

9 Standardised Protocols and Capabilities

This clause introduces a list of standardised protocols and capabilities that shall be supported by UMTS for the control and creation of services. The access protocols and the execution environment described below are essential for UMTS.

9.1 Access protocols

The access protocols shall allow the support of multimedia services. These services are characterised by the ability to dynamically change the number of participants and the number of connections during a call. The characteristics of the connections (confer the list of attributes used to describe a connection) may differ from one connection to another.

They are negotiated during call set-up. They may be independently and dynamically re-negotiated on application (the telecommunication requirements of the application changes) or network initiative (change of network load conditions, during a handover procedure) during the call.
The application may require synchronisation between some of the connections. TLater, this synchronisation shall not be lost during handover procedures.
Whenever a call is terminated in other types of networks, the negotiation shall take into account the limitations of these networks. Interworking shall be possible with PSTN, GSM, ISDN, B-ISDN and Internet networks. Later it shall also be possible to interwork with B-ISDN networks.
The access protocols shall allow a mobile station to have several calls active simultaneously .

9.2 Execution Environment

The execution environment is a set of standardised capabilities that shall allow the support of service provider/network operatorHE/SN specific services (i.e. both applications, teleservices and supplementary services). The execution environment shall be distributed between the IC card, terminal and network nodes. The terminal and the serving network capabilities shall be the only limiting factor for the support of the services designed to run on the execution environment. The execution environment is composed of the following building blocks;

- A standardised content description language for support of NO/SP specific user interfaces (both for information output and user input). This is intended only for platforms which are terminals.
- A standardised procedural language for support of NO/SP specific scripts. This language shall be common to all types of platforms. The scripts could be used for e.g. improving the user interface, adding new features to the terminal like the latest version of a codec, controlling the execution of a service.
- Standardised application programming interfaces for opening platform resources and capabilities to the scripts written with the standardised procedural language. These interfaces would be platform type dependent. The interfaces shall include primitives for accessing to the basic control functions, as illustrated on the figures 5 and 6 below.

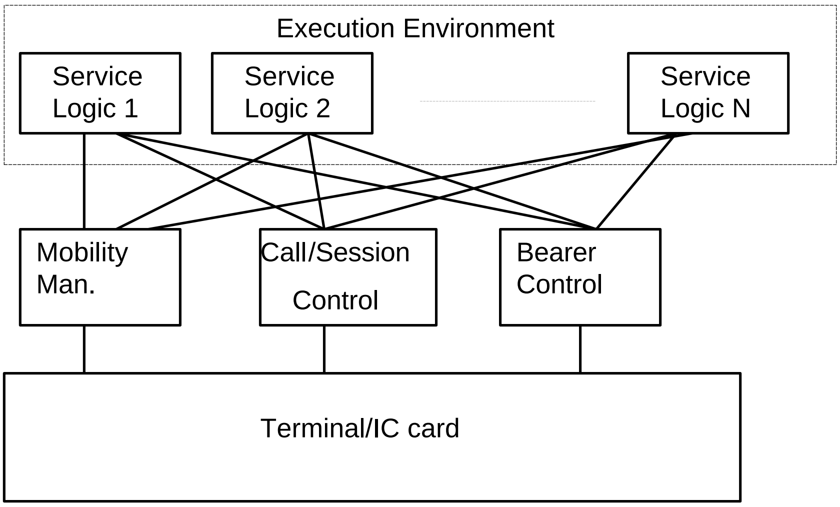


Figure 5: Execution Environment in the Mobile Station

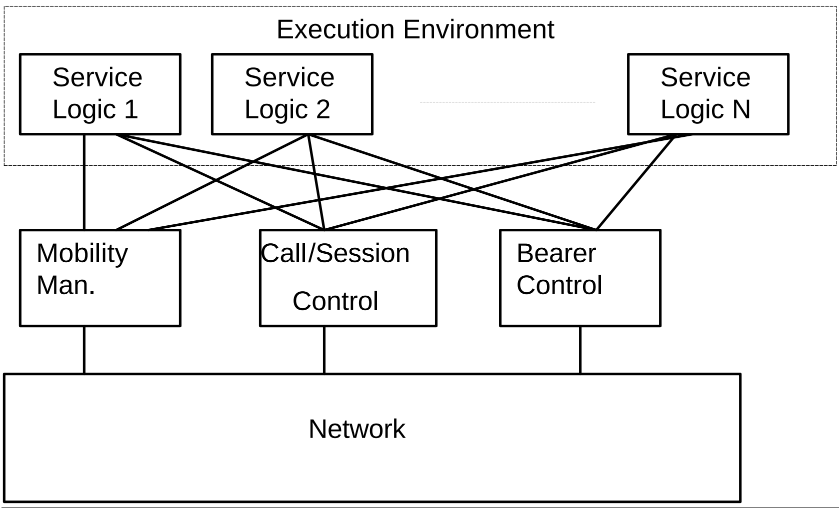


Figure 6: Execution Environment in the Network

- Call states, messages, information elements, values of information elements shall serve as triggers for subsequent interaction with service logic. The list of triggers is for further study and is likely to incorporate CAMEL, SIM Toolkit, MExE.
- Means to turn triggers on and off, and associate them with service logic shall be standardised.
- A standardised certification scheme and security model with several levels of trusts in order to control the scripts access rights to the platform resources and capabilities. This would be used to allow e.g. the SP and the SP only to access to USIM data.
- Standardised protocols for allowing the download of content description pages and scripts in the platform.

10 Existing GSM System features

Following GSM system features shall be supported by G-UMTS standards (for networks based on GSM evolution).

10.1 Network Identity and Time Zone (NITZ)

NITZ is specified in GSM 02.42.

10.2 Support of Localised Service Area (SoLSA)

SoLSA is specified in GSM 02.43.
Note : SoLSA modifications due to UTRAN related aspects are FFS.

10.3 Mobile station Execution Environment (MExE)

MExE is specified in GSM 02.57.

10.4 Location Services (LCS)

LCS is specified in GSM 02.71.
Note : LCS modifications due to UTRAN related aspects are FFS.

10.5 Customised Application for Mobile network Enhanced Logic (CAMEL)

CAMEL is specified in GSM 02.78.

10.6 Unstructured Supplementary Service Data (USSD)

USSD is specified in GSM 02.90
Note : USSD modifications due to UTRAN related aspects are FFS.

Annex A (informative): Examples of services built from service features

Call Barring

In standard GSM, the Call Barring services allow to prevent outgoing calls to certain sets of destinations, based on the number dialled and whether the user is roaming. In UMTS, it is proposed that this service allows to block outgoing calls based on a wider range of parameters which could include factors such as the time of day, day of week, location, type of call requested, cost of the service and/or destination. This would allow to develop Call Barring services tailored to business and personal markets to avoid abuse.

This service is invoked during the initial outgoing call set-up procedure and allow the call to be blocked prior to incurring any charges. This Service can be applied to any teleservice for both connection-oriented and connectionless-oriented services.

Call Filtering/Forwarding

In standard GSM, there is no call filtering service. All calls are presented to the user unless a call forwarding service is used to re-direct calls; there is no different call handling depending on the incoming call parameters (although differentiation on call type (voice/data) is possible).

In UMTS, the call filtering service allows the control of whether incoming calls are accepted, forwarded or terminated. The parameters which can be used to determine the final destination of a call may include the caller ID (CLI), original number dialled, time of day, current user location/network, user profile settings and current state of the terminal.

This service shall be two-stage; immediate call filtering (handled regardless of whether the terminal is online or not) and late call filtering (handled only if the terminal is online). It shall be possible to create and operate new call filtering services which can access any of the key parameters to handle calls in this way.

Hold

This service allows an established call to be maintained, whilst suspending use of the bearer from the incoming access point of the network. This saves on both air interface and network traffic resources when a call is temporarily suspended. The incoming access point in the network means either the originating UMTS terminal, or interworking point with another network.

Transfer

This service allows either an established or held call to be redirected to another destination. This may either be used by setting up a new call to the destination first, or simply redirecting the existing call to the new destination. It shall be possible to revert such a call back to the diverting terminal at any time before it is accepted (answered) by the new destination. The UMTS system shall ensure that an optimal traffic route is used after the call has been answered by its new (final) destination.

Call-back When Free

This service can be invoked where a call (or a connectionless message) cannot be delivered to its destination because it is in use. The UMTS system will inform the requesting entity when the destination is next able to accept the call, allowing a new call to be originated. This allows existing GSM services, such as Call-back When Free to be implemented. Where multiple requests are outstanding for a terminal which becomes available, the system will determine in which order the requests are handled, probably in a serial manner. Ideally, it shall be possible to create the service logic which determines the order used from a range of accessible parameters.

Annex B (informative):
Change history

Annex B (informative) :
Description and analysis of communication schemes

This annex gives a high level classification and description of communications requirements from end users and applications.

B.1 Communication schemes

The requirements on bearer services are based on an analysis of user and application needs. Four end-user groups are identified according to four distinctly different communication schemes; *background traffic, interactive traffic, real time streams, real time conversation*. For each scheme one (or two) fundamental characteristic(s) for QoS is identified and the resulting overall requirement(s) is derived. Of course, when the requested service/application bearer requirements are converted into bearer service attributes (as defined in section 5.2), these fundamental characteristics are reflected in the values allocated to the bearer service attributes.

B.1.1 Background traffic

When the end-user, that typically is a computer, sends and receives data-files in the background, this scheme applies. Examples are background delivery of E-mails, SMS, download of databases and reception of measurement records.

Background traffic is one of the classical data communication schemes that on an overall level is characterised by that the destination is not expecting the data within a certain time. The scheme is thus more or less delivery time insensitive. Another characteristic is that the content of the packets must be transparently transferred (with low bit error rate).

Background traffic - fundamental characteristics for QoS:

- the destination is not expecting the data within a certain time
- preserve payload content

The resulting overall requirement for this communication scheme is to support non-real time services without any special requirement on delay.

B.1.2 Interactive traffic

When the end-user, that is either a machine or a human, is on line requesting data from remote equipment (e.g. a server), this scheme applies. Examples of human interaction with the remote equipment are: web browsing, data base retrieval, server access. Examples of machines interaction with remote equipment are: polling for measurement records and automatic data base enquiries (tele-machines).

Interactive traffic is the other classical data communication scheme that on an overall level is characterised by the request response pattern of the end-user. At the message destination there is an entity expecting the message (response) within a certain time. Round trip delay time is therefore one of the key attributes. Another characteristic is that the content of the packets must be transparently transferred (with low bit error rate).

Interactive traffic - fundamental characteristics for QoS:

- request response pattern
- preserve payload content

NOTE: Interactive applications might differ between interactive games and interactive data transfer such as web browsing.

The resulting overall requirement for this communication scheme is to support interactive non-real time serviceswith low round-trip delay.

B.1.3 Real time streams

When the user is looking at (listening to) real time video (audio) the scheme of real time streams applies. The real time data flow is always aiming at a live (human) destination. It is a one way transport.

This scheme is one of the newcomers in data communication, raising a number of new requirements in both telecommunication and datacommunication systems. First of all it is a unidirectional stream with high continuous utilisation (i.e. having few idle/silent periods.) It is also characterised by that the time relations (variation) between information entities (i.e. samples, packets) within a flow must be preserved, although it does not have any requirements on low transfer delay.

The delay variation of the end-to-end flow must be limited, to preserve the time relation (variation) between information entities of the stream. But as the stream normally is time aligned at the receiving end (in the user equipment), the highest acceptable delay variation over the transmission media is given by the capability of the time alignment function of the application. Acceptable delay variation is thus much greater than the delay variation given by the limits of human perception.

Real time streams - fundamental characteristics for QoS:

- unidirectional continuous stream
- preserve time relation (variation) between information entities of the stream

The resulting overall requirement for this communication scheme is to support streaming real time services having unidirectional data flows with continuous utilisation. (There are less hard requirements on delay and packet loss ratio, i.e. the ratio of lost or corrupted packets out of all packets sent.)

B.1.4 Real time conversation

The most wellknown use of this scheme is telephony speech (e.g. GSM), but with Internet and multimedia a number of new applications will require this scheme, for example voice over IP and video conferencing tools. Real time conversation is always performed between peers (or groups) of live (human) end-users. This is the only scheme where the required characteristics are strictly given by human perception (the senses). Therefore this scheme raises the strongest and most stringent QoS requirements.

The real time conversation scheme is characterised by that the transfer time must be low because of the conversational nature of the scheme and at the same time that the time relation (variation) between information entities of the stream must be preserved in the same way as for real time streams. The maximum transfer delay is given by the human perception of video and audio conversation. Therefore the limit for acceptable transfer delay is very strict, as failure to provide low enough transfer delay will result in unacceptable lack of quality. The transfer delay requirement is therefore both significantly lower and more stringent than the round trip delay of the interactive traffic case.

Real time conversation - fundamental characteristics for QoS:

- preserve time relation (variation) between information entities of the stream
- conversational pattern (stringent and low delay)

The resulting overall requirement for this communication scheme is to support conversational real time services with low transfer delay as given by the human perception. (There are less hard requirements on packet loss ratio.)

B.2 Adaptability and bearer service negotiation

Applications using the *interactive* or *real time conversational* communication schemes can also be described according to their possibilities for adapting to different environmental conditions as follows:

- Rigid applications; these applications can not adapt at all (e.g. GSM full rate speech.)
- Adaptive applications; these applications can adapt to the environment; they therefore require the network to support service negotiation. (e.g. multi-rate speech codecs)
- Elastic applications; these applications adapt totally to the environment and do therefore not require service negotiation (e.g. web browsing.)

The resulting overall requirement is to support service negotiation.

Annex C (informative) : Change history

Change history					
SMG No.	**TDoc. No.**	**CR. No.**	**Section affected**	**New version**	**Subject/Comments**
SMG#25				3.0.0	Approved at SMG#25 Sophia Antipolis 17-20 March 1998
SMG#26	98-0325	A001	Section 3.1 (a new definition is added) and section 5 (the current text is clarified).	3. 0.11.0	Clarification of the sections where the bearer services are characterised.
Pre-SMG#28	SMG1 Tdoc 98-0864	A002	3.2, 4.1, 4.3, 4.4, 6.1, 6.4.1, 6.4.3, 6.4.4., 6.4.5, 8.2, 8.3, 8.5, 8.7, 8.8, 9.1, 9.2	Draft 3.2.0	SMG1 Agreed at this stage Aligning this specification with 22.00
Pre-SMG#28	SMG1 Tdoc 98-0870	003	Sections 2, 9	Draft 3.2.0	SMG1 Agreed at this stage References to relevant GSM specs has been added.
Pre-SMG#28	SMG1 Tdoc 0895 (865#5)	004	Annex B	Draft 3.2.0	SMG1 Agreed at this stage In line with views expressed by SMG2 and SMG12 about descriptions and analysis of communication schemes.
Pre-SMG#28		005	Section 8	Draft 3.2.0	Service features are only used to create services (as building blocks) and not to modify and delete services. (Added Jan 27, 1999)
SMG#28				Version 3.2.0	Approved Versions
SA#2 Florida		006	6.4	Version 3.3.0	Cell Broadcast Service in UMTS.

History

Document history	
June 1998	Version 3. 0.11.0 Unpublished
January 1999	**v.3.2.0 - with 43 CRs accepted by SMG1 - Presented to TSG SA WG1 - To be presented to SMG#28 for Approval**
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~~5.2. Bearer services~~

~~UMTS phase 1 shall support GSM phase 2+ Release '99 data bearer services :~~

~~***Circuit switched data*: Circuit switched data services and "real time" data services shall be provided for interworking with the PSTN/ISDN so that the user is unaware of the access network used (UMTS and GSM access network or handover between access networks). Both transparent (constant delay) and non-transparent (zero error with flow control) services shall be supported. These data services shall operate with minimum loss of data on handover between the GSM access network and the UTRAN.~~

~~***Packet switched data*: Packet switched data services shall be provided for interworking with packet networks such as IP-networks and LANs. The standard shall provide mechanisms which ensure the continuity of packet based services upon handover e.g. between GSM and UMTS.~~

~~5.3 Emergency call~~

~~UMTS phase 1 shall support following requirements on emergency call :~~

~~It should be possible to connect an emergency call to an appropriate destination (i.e. police station, fire brigade etc.) in the countries/regions where more than one emergency destinations are provided, e.g. according to the dialled digits. In Japan, for example, “110” is used for police and “119” for fire brigade.~~