ETSI TS 122 001 V3.2.0 (2000-03)

Technical Specification

Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); Principles of circuit telecommunication services supported by a Public Land Mobile Network (PLMN) (3G TS 22.001 version 3.2.0 Release 1999)

GLOBAL SYSTEM FOR MOBILE COMMUNICATIONS

Reference

RTS/TSGS-0122001UR1

Keywords

GSM, UMTS

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Important notice

Individual copies of the present document can be downloaded from:
http://www.etsi.org

The present document may be made available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be the printing on ETSI printers of the PDF version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at http://www.etsi.org/tb/status/

If you find errors in the present document, send your comment to:
editor@etsi.fr

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (http://www.etsi.org/ipr).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

This Technical Specification (TS) has been produced by the ETSI 3rd Generation Partnership Project (3GPP).

The present document may refer to technical specifications or reports using their 3GPP identities, UMTS identities or GSM identities. These should be interpreted as being references to the corresponding ETSI deliverables.

The cross reference between GSM, UMTS, 3GPP and ETSI identities can be found under www.etsi.org/key .

Contents

Foreword

This Technical Specification (TS) has been produced by the 3rd Generation Partnership Project (3GPP).

The present document defines the telecommunication services supported by a GSM PLMN within the digital cellular telecommunications system (Phase 2+).

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

Version x.y.z

where:

- x the first digit:
  - 1 presented to TSG for information;
  - 2 presented to TSG for approval;
  - 3 or greater indicates TSG approved document under change control.
- y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.
- z the third digit is incremented when editorial only changes have been incorporated in the document.

0 Scope

The present document covers the definition of the circuit telecommunication services supported by a PLMN. The purpose of the present document is to provide a method for the characterization and the description of these telecommunication services.

TS 22.101 describes overall service principles of a PLMN.

0.1 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.

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

[2] ITU-T Recommendation I.221: "Common specific characteristics of services".

[3] ITU-T Recommendation X.200: "Information technology - Open Systems Interconnection - Basic reference model: The basic model".

[4] TS 22.101: "UMTS Service Principles".

[5] TS 22.002: "Bearer services supported by a PLMN".

[6] TS 22.003: "Teleservices supported by a PLMN".[7] TS 22.004: "General on Supplementary Services".

[8] TS 27.001: "General on Terminal Adaptation Functions (TAF) for Mobile Stations (MS)".

[9] TS 22.030: "Man-Machine Interface (MMI) of the User equipment (MS)".

[10] TS 22.081: "Line Identification Supplementary Servicess; Stage 1".

[11] TS 22.135: "Multicall; Stage 1".

[12] TR 21.905: "Vocabulary for 3GPP Specifications".

[13] GSM 04.08: "Digital cellular telecommunications system (Phase 2+); Mobile radio interface layer 3 specification".

0.2 Abbreviations

Abbreviations used in the present document are listed in GSM 01.04 and TR 21.905 [12].

1 Framework for the description of telecommunication services

1.1 The attribute method of characterization of circuit telecommunication services

This characterization is made by using a set of attributes. A telecommunication service attribute is a specific characteristic of that service whole values distinguish it from other telecommunication services. Particular values are assigned to each attribute when a given telecommunication service is described and defined.

A list of definitions of attributes and values used for bearer services and teleservices is contained in, respectively, annex A and annex B.

2 Description of circuit telecommunication services by the attribute method

2.1 General

Telecommunication services are described by attributes which define service characteristics as they apply at a given reference point where the customer accesses the service. The description of a telecommunication service by the method of attributes is composed of:

- technical attributes as seen by the customer; and
- other attributes associated with the service provision, e.g. operational and commercial attributes.

2.2 Categorisation of telecommunication services

The concepts introduced in the present document are illustrated in table1.

Table 1: Categorisation of telecommunication services

TELECOMMUNICATION SERVICES			
BEARER SERVICE		TELESERVICE	
Basic Bearer Service	Basic Bearer service + supplementary services	Basic Teleservice	Basic Teleservice + supplementary service

3 Characterization of circuit telecommunication services

3.1 General

A telecommunication service supported by a PLMN is characterized and described by service attributes.

There are two groups of service attributes applicable to user information flow:

- low layer attributes;
- high layer attributes.

Bearer services are characterized only by low layer attributes. Teleservices are characterized by both low layer attributes and high layer attributes.

The basic characteristics of a telecommunication service are described by the basic service attributes.

The additional characteristics associated with a supplementary service which modify or supplement a basic telecommunication service are described in Specification TS 22.004 [7].

3.2 Bearer services

Bearer services are characterized by a set of low layer attributes in Specification GSM 02.02. These attributes are classified into four categories:

- information transfer attributes;
- access attributes;
- interworking attributes;
- general attributes, including operational and commercial attributes.

The bearer capability defines the technical features of a bearer service as they appear to the user at the appropriate access point. For the time being, the bearer capability is characterized by information transfer, access and interworking attributes. A bearer capability is associated with every bearer service.

3.3 Teleservices

Circuit teleservices provide the full capacity for communication by means of terminals and network functions and possibly functions provided by dedicated centres.

Circuitteleservices are specified in TS 22.003 [6].Teleservices are characterized by a set of low layer attributes, a set of high layer attributes and operational and commercial attributes.

Low layer attributes are those used to characterize the bearer capability. High layer attributes are used in Specification TS 22.003 [6] to describe high layer (i.e. layer 4-7) information transfer related characteristics. They refer to functions and protocols of layers 4-7 in the ITU-T Recommendation X.200 framework which are concerned with the transfer, storage and processing of user messages (provided by a subscriber's terminal, a retrieval centre or a network service centre).

Therefore, not all attributes can be applied directly at the user to terminal interface as they represent two kinds of features, the bearer capability and the terminal features, that are not directly perceived by the user.

4 Provision of telecommunication services

Specifications GSM 02.10 and GSM 02.11 define some aspects of the provisions of telecommunication services by a GSM PLMN.

The provision of telecommunication services implies:

- subscription of basic services and possibly subscription to supplementary services;
- registration into a service directory;
- compatibility between terminals;
- interworking capabilities (see GSM 09 series of specifications).

The user's subscription to a Basic or Supplementary service is normally verified by the network prior to completion of Call Establishment and/or Supplementary Service operation. This subscription checking shall be performed in accordance with the following subclauses.

4.1 Subscription checking for Basic Services

General

Subscription checking is the function/process to ascertaining whether a subscriber has the authorization to use the particular Basic Service deduced from the call set-up parameters. It is the responsibility of the HPLMN to transfer, to the VPLMN, only the subscription data corresponding to those services a given subscriber is entitled to use in that VPLMN.

For mobile originated calls, subscription checking is performed in the VLR, whilst for mobile terminated calls it is performed in either the HLR or the VLR (determined as described below). The prerequisite for executing the subscription check is a successful deduction of a Basic Service from the Compatibility Information contained in the call set up, i.e. Bearer Capability Information Element and, in some cases, also the Low Layer and High Layer Compatibility Information elements.

For mobile originated calls an UE shall indicate the requested service by appropriate compatibility information elements according to GSM 27.001 [8]. This information is mapped to an individual Basic Service code (i.e. the MAP representation) by the MSC in order to be compared with the subscriber data available in the VLR.

An equivalent process is required in the HLR for mobile terminated calls, where the caller's requested service is indicated to the HLR (by the ISDN) by exhaustive compatibility information consisting of ISDN Bearer Capability Information Elements and in some cases - depending on the service requested - also of Low Layer and High layer Compatibility information elements. In case the compatibility information is not exhaustive, e.g. when the call is originated/transited by a PSTN, no Basic Service can be deduced and subscription checking cannot be performed in the "normal" way. Instead, rules for the Single and Multi Numbering Schemes apply.

In the Multi Numbering Scheme the Basic Service can be deduced by information stored in the HLR against the called number and hence an implicit subscription check is performed. In the Single Numbering Scheme, the Basic Service cannot be deduced until the UE has responded to the set up and therefore the HLR cannot perform subscription check. Instead, the VLR/MSC will perform the subscription check or calls are passed "unfiltered" (as regards subscription check), at the network operators' discretion.

Bearer Services

GSM 02.02 lists the Bearer Services, each of them with a specific "BS number". Single services defined independent of the fixed network user rate are called General Bearer Services. These distinct [numbered] services may individually be provided to a subscriber.Whichever the subscription arrangements are, all PLMNs (MSCs, VLRs and HLRs) shall be able to allow - as regards subscription checking - the use of individually subscribed-to Basic Services, within the range of services supported by the PLMN. That is, whenever it is possible to deduce the Basic Service from a call set up, subscription check shall be performed at the granularity of that particular Basic Service or the group to which it belongs.

TeleServices

TS 22.003 [6] lists the TeleServices, each of them with a specific "TS number". These may be provided to subscribers individually or combined, to the operators' discretion, however TS 12 (Emergency calls) and TS 23 (CBS) are not subscribable. But, as for Bearer Services, networks shall be able to handle subscription checking at the granularity of individual TeleServices.

Table 2 summarizes the basis on which a successful subscription checking will result. It also describes on which basis Supplementary Service handling for a given call set-up should be performed.

Table 2

Set Up	Subscription Check	SS handling
BS 20	BS 20	BS Group 2x
BS 30	BS 30	BS Group 3x
TS 11	TS 11, TS Group 1x or TS Group All	TS Group 1x
TS 12	N.A.	
TS 21	TS 21, TS Group 2x or TS Group All	TS Group 2x
TS 22	TS 22, TS Group 2x or TS Group All	TS Group 2x
TS 23	N.A.	
TS 61	TS 61, TS Group 6x or TS Group All	TS Group 6x
TS 62	TS 61, 62, Group 6x or TS Group All	TS Group 6x
TS 91	TS 91, TS Group 9x or TS Group All	TS Group 9x
TS 92	TS 92, TS Group 9x or TS Group All	TS Group 9x
Legend: - set-up: The Basic Service which is set up for the call; - subscription check: Required VLR or HLR data for successful subscription check; - SS handling: Against which VLR or HLR data SS handling should be performed. For example; a call set-up indicating BS61 and Asynchronous mode should be treated for SS purposes in accordance with the SS-data stored against BS group 2x.		

When TS61 is requested in a call set-up and the subscription check for TS61 is negative, but a subscription check for TS62 is positive, then the call shall proceed according to the TS 22.003 [6]and TS 27.001 [8]. If a subscription check for both TS61 and TS62 is negative, then the call shall be released.

4.2 Subscription checking for Supplementary Services

This is described in GSM 02.04 and the GSM 03.8x series of specifications.

Annex A (normative): List of definition of attributes and values used for bearer services

A.1 Information transfer attributes

A.1.1 Information transfer capability

This attribute describes the capability associated with the transfer of different types of information through a PLMN and another network or through a PLMN.

Values:

- unrestricted digital information;
  transfer of information sequence of bits at its specified bit rate without alteration; this implies bit sequence independence, digit sequence integrity and bit integrity.
- speech;
  digital representation of speech information and audible signalling tones of the PSTN coded according to the encoding rule defined in the GSM 06 series of specifications.
- 3,1 kHz Ex PLMN;
  unrestricted digital information transfer within the PLMN and 3.1 kHz audio restricted within the ISDN.
- group 3 Fax;
  transfer of Group 3 Fax information.

A.1.2 Information transfer mode

This attribute describes the operational mode of transferring (transportation and switching) through a PLMN.

Values:

- circuit.

A.1.3 Information transfer rate

This attribute describes the bit rate (circuit mode. It refers to the transfer of digital information between two access points or reference points.

Values:

- appropriate bit rate, throughput rate.

A.1.4 Structure

This attribute refers to the capability of the PLMN and if involved other networks to deliver information to the destination access point or reference point in a structure NOTE: This attribute has not been utilised in TS 22.002 [5] or TS 22.003 [6].

Values:

- not applicable.

A.1.5 Establishment of communication

This attribute associated with a telecommunication service describes the mode of establishment used to establish and a given communication.

In every telecommunication service communication may be between users within the PLMN or between a user in the PLMN and a user in another network.

Values:

- demand Mobile Originated (MO) only;
- demand Mobile Terminated (MT) only;
- demand Mobile Originated or Terminated (MO, MT).

A.1.6 Communication configuration

This attribute describes the spatial arrangement for transferring information between two or more access points. It completes the structure associated to a telecommunication services as it associates the relationship between the access points involved and the flow of information between these access points.

Values:

- point-to-point communication;
  this value applies when there are only two access points.
- multipoint communication;
  this value applies when more than two access points (1) are provided by the service. The exact characteristics of the information flows must be specified separately based on functions provided by the PLMN.

NOTE 1: The number of access points can be undefined.

- broadcast communication;
  this value applies when more than two access points (2) are provided by the service. The information flows are from a unique point (source) to the others (destination) in only one direction.

NOTE 2: The number of destination access points can be undefined.

A.1.7 Symmetry

This attribute describes the relationship of information flow between two (or more) access points or reference points involved in a communication.

It characterizes the structure associated to a communication service.

Values:

- unidirectional;
  this value applies when the information flow is provided only in one direction.
- bidirectional symmetric;
  this value applies when the information flow characteristics provided by the service are the same between two (or more) access points or reference points in the forward and backward directions.
- bidirectional asymmetric;
  this value applies when the information flow characteristics provided by the service are different in the two directions.

A.1.8 Data compression

This attribute indicates whether use of a data compression function is desired (and accepted) between an MT and IWF.

Values:

- use of data compression requested/not requested;
- use of data compression accepted/not accepted.

A.2 Attributes describing the access at the user equipment

A.2.1 Signalling access

This attribute characterized the protocol on the signalling channel at a given access point or reference point Values:

- manual;
- appropriate V-series protocol;
- appropriate X-series protocol;
- I-series stack of signalling protocols.

A.2.2 Information access

A.2.2.1 Rate

This attribute describes either the bit rate (circuit mode including transparent access to a PSPDN) or variable bit rate (packet mode) used to transfer the user information at a given access point or reference.

Values:

- appropriate bit rate;
- variable bit rate.

A.2.2.2 Interface

This attribute describes the interface according to the protocol used to transfer user information at a given access point or reference.

Values:

- appropriate V-series DTE/DCE interface;
- appropriate X-series interface;
- S interface;
- analogue 4-Wire interface.

A.3 Interworking attribute

A.3.1 Type of terminating network

Communication can be established between a UE in a PLMN (originating network) and a terminal in a network (terminating network) including the same PLMN or another PLMN. The attribute designates the terminating network.

NOTE 1: The terms "originating" and "terminating" do not indicate the direction of communication establishment.

NOTE 2: This attribute does not reflect whether there is none, one or several transit networks between the originating and terminating networks.

Values:

- PSTN;
- ISDN;
- PSPDN;
- PDN;
- PLMN;
- direct access networks.

A.3.2 Terminal to terminating network interface

This attribute describes the interface between a terminal equipment and the terminating network.

Values:

- appropriate V-series (DTE/DCE) interface;
- appropriate X-series interface;
- analogue 2 resp. 4 wire interface;
- S interface (D+B+B).

A.4 General attributes

A.4.1 Supplementary services provided

This attribute refers to the supplementary services to a given telecommunication service.

Values:

- appropriate supplementary services.

A.4.2 Quality of service

The Bearer Services use the Quality of Service attribute to indicate one of the following values:

- transparent;
  service characterized by constant throughput, constant transit delay and variable error rate.
- non-transparent;
  service characterized by an improved error rate with variable transit delay and throughput.

A.4.3 Commercial and operational

A.4.4 Service interworking

Annex B (normative): List of definitions of attributes and values used for teleservices

B.1 High layer attributes

B.1.1 Type of user information

This attribute describes the type of information which the communication offered to the user by the teleservice is based on.

Values:

- speech;
- short message;
- facsimile.

B.1.2 Layer 4 protocol functions

B.1.3 Layer 5 protocol functions

B.1.4 Layer 6 protocol functions

B.1.5 Layer 7 protocol functions

B.2 Low layer attribute (bearer capabilities)

The low layer attributes describe the bearer capabilities which support the teleservice. These low layer attributes and their values are the same as presented in Annex A: List of definitions of attributes and values used for bearer services.

B.3 General attributes

The general attributes are the same as presented in annex A: List of definitions and values used for bearer services.

Annex C (normative): Definition of "busy" in a PLMN

C.1 Scope

This annex describes the conditions under which a given mobile subscriber (station) is considered as "busy". In general, this occurs whenever the resources associated with that UE (and needed to successfully complete the call) exist but are not available for that call. The description is based on the busy definition in the ISDN (CCITT Recommendation I.221).

In addition, the operation of some Supplementary Services occurs when certain of these resources are busy. Therefore, these "resources busy" are also described herein.

This annex does not cover the cases, when network resources not associated with a given destination are unavailable, or when such resources are out-of-service or otherwise non-functional.

C.2 Network Determined User Busy (NDUB) condition

This condition occurs, when a call is about to be offered, if the information (i.e. traffic) channel is busy and the maximum number of total calls has been reached (see note).

This condition also occurs, when a call is about to be offered and an already on-going call attempt (incoming or outgoing) is in the establishing phase, i.e. not yet active.

When NDUB condition occurs, the PLMN will clear the call and indicate "busy" back towards the calling subscriber (see also clause 4).

NOTE: The value of the maximum number of calls is 1 for the basic call. When the supplementary service "Call Waiting" is applicable the value is n+1 where n is the maximum number of calls that can be waiting.

TS 22.135 [11] defines NDUB for Multicall environment.

C.3 User Determined User Busy (UDUB) condition

This condition occurs when a call is offered to a user equipment and the UE responds "user busy" because the subscribers resources (terminal or person using them) are busy. Then the PLMN will clear the call with the indication "busy" back towards the calling subscriber (see also clause 4).

C.4 Mobile subscriber busy

A mobile subscriber is considered to be busy if either a "Network Determined User Busy" or a "User Determined User Busy" condition occurs.

Some supplementary services (e.g. Call Forwarding on Busy) may cause the call not to be cleared when a busy condition occurs.

Annex D (normative): Call set-up procedures

D.1 Scope

This annex specifies the service requirements for call set-up, both Mobile originated and mobile terminated, in a network, including the establishment of radio contact.

D.2 Mobile Originated Call Set-up

When an UE wishes to start a call and there is no existing radio connection, it requests a signalling channel. When such a signalling channel has been allocated to the UE, the UE can transfer the call set-up information.

A traffic channel may be allocated at any time before the network informs the UE that the remote user has answered.

For a call to be set up, certain information needs to be sent by the UE to the network, defining the call. This information may be provided as default by the MS, it may be derived from the SIM or be entered by the user either directly into the UE or from a DTE by using the DTE/DCE Interface.

The following information is sent. Where necessary, default values will generally be inserted by the UE if not directly specified by the user. The Teleservice Emergency Calls are set up using a special procedure not using the fields described in this clause (except for the Bearer Capability).

D.2.1 Called Party Address

This is the address of the called party, generally as defined in GSM 03.03, using the TON/NPI specified below. In the case of Dedicated PAD or Packet Access, if NPI is set to PNP, the called party address field may be used to specify the profile to be used. In that case, the address of the called DTE will be given in-band as the second part of two-stage call set-up.

D.2.2 Calling/Called Party Sub-address

This is the sub-address of the calling/called party, as defined in GSM 03.03, in order to provide interworking with ISDN. This is described in more detail in ETS 300 059. Support and use of these fields are optional.

D.2.3 Type of Number

This indicates the format of the called party address. The selection procedure is given in TS 22.030 [9]. The following Types of Number are commonly used:

- International Format;
- Open Format ("Unknown");
- Dedicated PAD/Packet Access.

D.2.4 Number Plan Indicator

This indicates the number plan of the called party address. Either of the following number plans may be the "default", depending on the contents of the Called Party Address (see TS 22.030 [9]):

- ISDN/Telephony E.164;

- unknown.

Alternatively, one of these number plans may be specified if appropriate:

- data network X.121;
- telex network F.69;
- National Numbering Plan;
- Private Numbering Plan.

D.2.5 Bearer Capability

This is used to define the type of call to be set up (telephony, data, rate etc.) For most applications, the UE will use a set of default conditions, generally on the assumption of a telephony call, unless otherwise set. These may be overridden by the user (or DTE via the DTE/DCE Interface) if desired except for the determination of the channel mode (full or half rate, speech codec conversion).

The UE shall indicate to the network its channel mode capability in terms of the data channels and the speech codec versions supported.

The network decides which mode to use on the basis of the requested bearer or teleservice, the available network resources and the channel mode capability of the UE:

- for the "alternate" and "followed-by" services, the same principle applies (with the exception of TS61, where a Full Rate or an Enhanced Full Rate channel shall be provided);
- for the full set of parameters and values, refer to GSM 04.08;
- for data services see the GSM 07 series.

Lower Layer Compatibility and Higher Layer Compatibility Information Elements may also be included.

D.2.6 Calling Line Indication Restriction Override

If the user wishes to override the calling line identification restriction, he may indicate this on a per-call basis as described in TS 22.030 [9] and TS 22.081 [10].

D.2.7 Action of the Network on Call Set-up

On receipt of the call set-up message, the network shall attempt to connect the call. However, if insufficient information has been provided by the UE to indicate the exact Bearer Capability requirements (e.g. due to missing or optional values or for rate adaptation for data), the network may insert the missing information, if this is possible, and the call set-up shall proceed using the new information. If the call set-up is unsuccessful, the network shall notify the UE of the cause.

D.3 Mobile Terminated Call Set-up

Using the procedures described in TS 22.011, the network knows the location area where the UE is positioned. If the UE is not already in two way radio communication with the network, the network pages the MS. Upon receiving its page message, the UE establishes communication with the selected cell. The network then allocates a channel which is used for signalling and sends call set-up information to the UE.

A traffic channel may be allocated at any instant until just after the call is answered by the UE.

The network indicates to the UE that it wishes to offer the UE a call. This notification includes the proposed bearer capability information, where available (see subclause D.2.5).

D.3.1 Bearer Type

If the calling party specifies the required bearer capability this shall be used for the call set-up attempt. If the calling party does not specify the required bearer capability (e.g. because the call originated in the PSTN), the network shall attempt to determine the bearer capability to be used as described below.

The network may use a multi-numbering scheme to define the bearer capability by the MSISDN. In a multi-numbering scheme several MSISDNs are associated with one IMSI. Each MSISDN is used for a different bearer capability. If the network uses a multi-numbering scheme and the calling party has not specified the required bearer capability then the network shall use the bearer capability associated with the called party MSISDN.

The network may use a single-numbering scheme, in which one MSISDN is associated with each IMSI. If the network uses a single-numbering scheme and the calling party has not specified the required service then the network shall omit the bearer capability information.

D.3.2 Response of the UE

On receipt of the call set-up request from the network, the UE shall check that it is able to support the type of call requested and that it is not User Determined User Busy (see annex C). The UE then alerts the user.

If the UE is unable to support the type of call requested, or the information is incomplete, the UE shall, if possible and not restricted by requirements in other ETSs, reply to the network proposing an alternative set of parameters, indicating those that are different from those proposed by the network. The network then either accepts this new proposal or terminates the call attempt.

D.3.3 Description of Call Re-establishment

Call re-establishment allows the user equipment to attempt to reconnect a call following the loss of radio coverage between the UE and the network while a call is in progress. Call re-establishment may be initiated by the UE when it detects this situation, if supported in the network.

Call re-establishment is mandatory in the ME and optional in the network.

Annex E (normative): Automatic calling repeat call attempt restrictions

Call set up attempts referred to in this annex are assumed to be initiated from peripheral equipment or automatically from the MT itself.

A repeat call attempt may be made when a call attempt is unsuccessful for the reasons listed below (as defined in GSM 04.08 [12]).

These reasons are classified in three major categories:

1) "Busy destination":

- cause number 17 User busy.

2) "Unobtainable destination - temporary":

- cause number 18 No user responding;

  19 User alerting, no answer;

  27 Destination out of order;

  34 No circuit/channel available;

  41 Temporary failure;

  42 Switching Equipment congestion;

  44 Requested circuit/channel not available;

  47 Resources unavailable, unspecified.

3) "Unobtainable destination - permanent/long term":

- cause number 1 Unassigned (unallocated) number;

  3 No route to destination;

  22 Number changed;

  28 Invalid number format (uncompleted number);

  38 Network out of order.

NOTE 1: Optionally, it is allowed to implement cause number 27 in Category 3, instead of Category 2, as this is desirable already in Phase 1.

The table below describes a repeat call restriction pattern to any B number. This pattern defines a maximum number (n) of call repeat attempts; when this number n is reached, the associated B number shall be blacklisted by the MT until a manual re-set at the MT is performed in respect of that B number. When a repeat attempt to anyone B number fails, or is blacklisted, this does not prevent calls being made to other B numbers.

For the categories 1 and 2 above, n shall be 10; for category 3, n shall be 1.

call attempts	Minimum duration between Call attempt
Initial call attempt	-
1st repeat attempt	5 sec
2nd repeat attempt	1 min
3rd repeat attempt	1 min
4th repeat attempt	1 min
5th repeat attempt	3 min
nth repeat attempt	3 min

The number of B numbers that can be held in the blacklist is at the manufacturers discretion but there shall be at least 8. However, when the blacklist is full the MT shall prohibit further automatic call attempts to any one number until the blacklist is manually cleared at the MT in respect of one or more B numbers.

When automatic calling apparatus is connected to an MT1 or MT2, or where an MTO is capable of auto-calling, then the MT shall process the call requests in accordance with the sequence of repeat attempts defined above, i.e. requests for repeat attempts with less than the minimum allowed duration between them shall be rejected by the MT.

A successful call attempt to a number which has been subject to the call restrictions shown above (i.e. an unsuccessful call set up attempt has previously occurred) shall reset the "counter" for that number.

The "counter" for an unsuccessfully attempted B number shall be maintained in 24 hours or until the MT is switched off.

The automatic calling repeat call attempt restrictions apply to speech and data services.

NOTE 2: The restrictions only apply to unsuccessful Call Control activity, not to Radio Resource Management or to Mobility Management, so multiple attempts at radio channel access are not limited by this mechanism.

Annex F(informative): Procedures for call progress indications

F.1 General

Indications of call progress, such as ringing, engaged, unobtainable, and no radio channel, may in principle be verbal message, tones, displayed text or graphical symbols. Which combination of these applies may depend on the message, the UE and selection by the user or PLMN operator. However, verbal announcements will generally be reserved for situations which are peculiar to a mobile network, where users may be unfamiliar with any tone chosen to indicate conditions such as "call diversion" or "subscriber not available".

It may also be desirable to add comfort indications (e.g. tones, noise, music, clicks) while a call is being connected, since silence may cause an unfamiliar user to believe that nothing is happening.

Generally, on data calls, and on the data part of alternate speech/data or speech-followed-by-data calls, PLMN generated network tones and announcements should be muted.

F.2 Supervisory tones

F.2.1 General

Supervisory Tones, indicating primarily ringing, engaged and unobtainable numbers, may be generated by both the PLMN and PSTN.

Except for ring tone, all tones indicating call progress to a user shall be generated in the UE, on the basis of signals from the network where available, and are according to the standard defined in the present document.

Tones sent to a caller to a UE will be generated in the network, generally local to the caller, and will be to the standard of his local exchange, except for mobile to mobile calls, where the tones will be generated in the calling UE. For mobile terminated calls, the ring tone will be generated in the called MSC (except OACSU).

F.2.2 Method

In the interests of early release of the traffic channel on failure to succeed in setting up a (mobile originated) call, where possible supervisory tones should be indicated over signalling channels. The UE will then generate the required tones. However, if the network generates an in-band announcement this will be indicated to the UE. In this case the UE shall connect the user to the announcement until instructed to release the call, either by the user or by the network. An alternate procedure may apply for UE able to generate appropriate announcements internally.

The ring tone will be sent over the traffic channel, since this channel must be available for traffic immediately it is answered (exception: Off Air Call Set Up). The Ring Tone is therefore generated by the PLMN or PSTN supporting the called phone.

On failed mobile terminated call attempts, the called MSC will either signal to the caller, if this is possible, or else will generate the required supervisory tones.

"Alert" is not a supervisory tone. The indication is signalled, and the UE may generate any form of indication to the user that the UE is being called.

F.2.3 Standard tones

UE generated tones will be generally in accordance with CEPT (GSM), ANSI T1.607 (PCS 1900), or Japan recommendations, where appropriate, and are listed in table 1. Any network originated tones will be according to PLMN or PSTN choice.

F.2.4 Applicability

This method will apply in all cases where signalling is capable of indicating the supervisory tone required. However, for connection to certain fixed networks where this signalling is not possible, fixed network tones will be carried over the traffic channel.

User equipment may employ any suitable technique to indicate supervisory information. However, if tones are employed, they shall be in accordance with the present document. The use of these tones in the MSC is preferred.

NOTE 1: The tones and/or announcement to the calling party should not be provided if the Information transfer capability is set to UDI.

NOTE 2: For a call with information transfer capability set to 3.1 kHz, the use of tones and/or announcement may cause the expiry of an awaiting answer timer in a modem or fax machine.

F.2.5 Comfort tones

If desired by the PLMN operator, the network may optionally introduce "comfort tones" while the call is being connected, during what would otherwise be silence. This would be overridden by indication of a supervisory tone, an announcement or by traffic. PLMNs may offer this feature optionally to incoming or outgoing callers.

The "comfort tones" may take the form of tones, clicks, noise, music or any other suitable form, provided that they cannot be confused with other indications that might be expected.

This feature is intended to indicate to the user that his call is progressing, to prevent him terminating the call prematurely.

Table 1: Supervisory tones in UEs

Tone		Frequency			Tolerance		Type		
		CEPT	**ANSI**	**Japan**	**CEPT ANSI**	**Japan**	**CEPT**	**ANSI**	**Japan**
1	Dial tone (optional)	425 Hz	350 Hz added to 440 Hz	400 Hz	15 Hz	20Hz	Conti- nuous	Continuous	Conti- nuous
2 *	Subscriber Busy (Called Number)	425 Hz	480 Hz added to 620 Hz	400 Hz	15 Hz	20Hz	Tone on 500ms Silence 500ms	Tone on 500ms Silence 500ms	Tone on 500ms Silence 500ms
3 *	Congestion	425 Hz	480 Hz added to 620 Hz	Optional	15 Hz	Optional	Tone on 200ms Silence 200ms	Tone on 250ms Silence 250ms	Optional
4	Radio Path Acknowledgement (Mobile Originated only) (optional)	425 Hz	425 Hz	400 Hz	15 Hz	20 Hz	Single tone 200ms	Single tone 200ms	Tone on 1 Sec Silence 2 Sec
5	{Radio Path Not Available {Call Dropped – Mobile originated only	425 Hz	425 Hz	Optional	15 Hz	Optional	200ms} On/off 200ms} for 3 burst	200ms} On/off 200ms} for 3 burst	Optional
6 *	Error/Special Information} Number Unobtainable } Authentication Failure }	950 Hz 1400 Hz 1800 Hz	950 Hz 1400 Hz 1800 Hz	Optional	50 Hz 50 Hz 50 Hz	Optional	{Triple Tone {Tones on 330ms {Silence 1.0s	Triple Tone {Tones on 330ms {Silence 1.0s	Optional
7	Call Waiting Tone (CEPT)	425 Hz (tolerance 15 Hz), on for 200 ms, off for 600 ms on for 200 ms, off for 3 s, on for 200 ms, off for 600 ms on for 200 ms. This tone is superimposed on the audio traffic received by the called user. Alternate tones are *acceptable* but not preferred.							
7	Call Waiting Tone (ANSI)	440 Hz, on for 300 ms, 9,7 s off followed by (440 Hz, on for 100 ms off for 100 ms, on for 100 ms, 9,7s off and repeated as necessary) This tone is superimposed on the audio traffic received by the called user.							
7	Call Waiting Tone (Japan)	Optional							
Definition of these and other tones, together with advice on announcements, may be found in CEPT T/CS 20-15 and in T/SF 23.									
NOTE: *: The duration of these tones is an implementation option. However, in each case, the UE should be returned immediately to the idle state, and will be able to originate/receive calls, which will override these tones.									
Ringing Tone (Alternative National options permitted)		425Hz	440 Hz added to 480 Hz	Optional	15 Hz	Optional	Tone on 1 s Silence 4 s	Tone on 2 s Silence 4 s	Optional
For application of Call Control Cause Information Elements to these tones, see F.4.									

F.3 Recorded announcements

In present networks, both fixed and cellular, the language of recorded announcements and displayed information is invariably that of the country of origin. However, this is generally undesirable in a multi-lingual environment such as is encountered on a global network with international roaming. It is therefore probably desirable to minimise the number of such announcements.

Advanced UEs may be designed which have the ability to generate announcements in the form desired by the user, e.g. in the language preferred by the user. In this case, it becomes necessary to block any verbal announcements sent from the network towards the UE, to avoid clashes with those generated by the UE. The UE may be allowed to block in-band announcements in case appropriate announcements according to the Cause Information Elements (F.3) can be generated. The default setting of the UE shall be "non blocking", which could be set by MMI command to "blocking".

Announcements generated by the PLMN and sent to callers to that PLMN will generally be in the language of the PLMN. However, on some fixed networks it will be possible for the message to be signalled back to the caller's local exchange, which will then generate the announcement in its local language.

F.4 Application of call control cause information elements to supervisory tones

The Cause Information Elements are listed and defined in GSM 04.08 [13]. This annex lists these elements and indicates which supervisory tone should be generated in response. It should be noted that some conditions (e.g. radio path not available, dropped call) may be deduced by the UE, rather than signalled explicitly over the air interface. All causes not listed below should result in the generation of tone 6. In case of multiple calls a tone should only be generated if it does not disturb an ongoing active call. "-" indicates no tone required.

Cause CC		Tone (see table 1)
16	Normal Clearing	1
17	User Busy	2
22	Number Changed	-
30	Response to STATUS ENQUIRY	-
31	Normal, unspecified	-
34	No circuit/channel available	3
41	Temporary Failure	3
42	Switching Equipment Congestion	3
44	Requested circuit/channel not available	3
49	Quality of Service Unavailable	3
58	Bearer Capability not available	3

Annex G (informative): Change history

Change history										
TSG SA#	**SA Doc.**	**SA1 Doc**	**Spec**	**CR**	**Rev**	**Rel**	**Cat**	**Subject/Comment**	**Old**	**New**
Dec 1999			02.01					Transferred to 3GPP SA1	8.1.0	3.0.0
SP-06	SP-99519	S1-991076	22.001	001		R99	D	Mainly an editorial update for GSM/3GPP use	3.0.0	3.1.0
SP-07	SP-000069	S1-000124	22.001	002		R99	D	Editorial modification for change of SMS-CB to CBS	3.1.1	3.2.0
SP-07	SP-000053	S1-000133	22.001	003		R99	C	Procedure for call progress indications	3.1.1	3.2.0

Version	Date	Information about changes
V3.0.0	December 1999	Transferred to TSG SA at 3GPP SA#6. Under TSG TSG SA Change Control.
V3.1.0	December 1999	Implemented CRs approved at SA #06.
V3.2.0	March 2000	Implemented CRs approved at SA #07.

History

Document history		
V3.2.0	March 2000	Publication