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SERIES E: OVERALL NETWORK OPERATION, TELEPHONE SERVICE, SERVICE OPERATION AND HUMAN FACTORS

Quality of telecommunication services: concepts, models, objectives and dependability planning – Terms and definitions related to the quality of telecommunication services

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Definitions, associated measurement methods and guidance targets of user-centric parameters for call handling in cellular mobile voice service

Recommendation ITU-T E.807

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Definitions, associated measurement methods and guidance targets of user-centric parameters for call handling in cellular mobile voice service

Summary

Call handling is an important aspect of cellular mobile voice service user experience. Call handling is executed end-to-end by the access and non-access stratum of the network (i.e., global systems for mobile communications (GSM), code division multiple access (CDMA) or universal mobile telecommunications system (UMTS)).

To enable regulators and operators measure call handling of a cellular mobile voice service for benchmarking and compliance, Recommendation ITU-T E.807 defines five parameters, describes the methodology in accessing them, and provides guidance targets.

These can benefit a regulator, stakeholder or any interested party to independently measure and report on delivered service user experience.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
1.0	ITU-T E.807	2014-02-13	12	11.1002/1000/12119

* To access the Recommendation, type the URL http://handle.itu.int/ in the address field of your web browser, followed by the Recommendation's unique ID. For example, http://handle.itu.int/11.1002/1000/11830-en.

FOREWORD

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The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at http://www.itu.int/ITU-T/ipr/.

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Definitions, associated measurement methods and guidance targets of user-centric parameters for call handling in cellular mobile voice service

1 Scope

This Recommendation provides an introduction into call handling in cellular mobile voice service systems, but most importantly relates signalling procedures to major key performance indicators used in the assessment of voice service handling. Some causative factors that incapacitate mobile networks not to perform at their desired levels in consonance with meeting minimum communication standards in respect of guidance targets are discussed herein. This Recommendation is intended to be a key reference document for both mobile network operators and regulators.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T E.800] Recommendation ITU-T E.800 (1994), *Terms and definitions related to quality of service and network performance including dependability*.

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following term defined elsewhere:

3.1.1 quality of service (QoS) [ITU-T E.800]: The collective effect of service performances, which determine the degree of satisfaction of a user of the service.

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 call set-up time or **voice service access time**: The period of time elapsing from the sending of a complete destination address (target telephone number) to the setting up of a call to the receiving terminal.

3.2.2 stand-alone dedicated control channel (**SDCCH**) or **radio resource control** (**RRC**) **congestion rate**: The probability of failure of accessing a stand-alone dedicated control or radio resource control channel during call set-up.

3.2.3 traffic channel congestion rate or **voice service non-accessibility ratio**: The probability of failure of accessing traffic channel(s) during call connections.

3.2.4 call drop rate or **voice service cut-off ratio**: The probability of a call terminating without the user's action.

3.2.5 call completion rate or **voice service retainability ratio**: The probability that a call, after being successfully set up, has to be maintained during a period of time, ending normally, i.e., according to the user's expectation.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

BS	Base Station
BSC	Base Station Controller
BSIC	Base Station Identity Code
BTS	Base Transceiver Station
CDMA	Code Division Multiple Access
GSM	Global Systems for Mobile communications
MOC	Mobile Originating Call
MTC	Mobile Terminating Call
MS	Mobile Station
MSC	Mobile Switching Centre
QoE	Quality of Experience
QoS	Quality of Service
RAB	Radio Access Bearer
RRC	Radio Resource Control
SDCCH	Stand-alone Dedicated Control Channel
TCH	Traffic Channel
TRX	Transceiver
UMTS	Universal Mobile Telecommunications System

5 Conventions

None.

6 List of QoE parameters

The following quality of experience (QoE) parameters have been identified as useful metrics for assessing the user experience of cellular mobile voice call handling.

Key performance indicators for mobile voice service

Parameter 1: call set-up time or voice service access time

Parameter 1 is the period of time elapsing from the sending of a complete destination address (i.e., target telephone number) to the setting up of a call to the receiving terminal.

Parameter 2: stand-alone dedicated control channel (SDCCH)/radio resource control (RRC) congestion rate

Parameter 2 is defined as the probability of failure of accessing a stand-alone dedicated control or radio resource control channel during call set-up.

Parameter 3: traffic channel congestion rate or voice service non-accessibility ratio

Parameter 3 is defined as the probability of failure of accessing traffic channel(s) during call connections.

Parameter 4: call drop rate or voice service cut-off ratio

Parameter 4 is the probability of a call terminating without the user's action.

Parameter 5: call completion rate or voice service retainability ratio

Parameter 5 is defined as the probability that a call, after being successfully set up, has to be maintained during a period of time, ending normally, i.e., according to the user's expectation.

7 QoE parameters with definitions, measures, guidance targets, guidelines on measurement/evaluation

Methodology

The methodology is based on three basic characteristics:

1. ***End-to-end measurements*** – Measurements reflect all aspects that impact the quality of a service.
2. ***Impartiality*** – Measurements are carried out under equal terms for operators using drive test equipment. Simultaneous measurements of different networks are performed, providing an accurate picture of how the networks perform under the same conditions, same time, at the same locations and with the same parameters, thus making it possible to perform comparative analysis of the observed performances. Measurements are done generically and do not require channel-locking or network-locking.
3. ***Objectivity*** – Tests are carried out in a totally automatic way, thus eliminating the subjectivity inherent to human intervention or decision.

Measurement profile

The measurement profile includes process standardization to guarantee the reliability of the test and the definition of testing parameters, thus making it possible to perform analyses and compare results.

Voice calls are performed in a series of two attempts within 10 seconds for a delay of 10 seconds between series. A successful call is to last a maximum of 60 seconds and has to be completed in a window of 90 seconds. The minimum time required for a call set up before the end of a call window is 30 seconds. The maximum call set-up time is 30 seconds. Guard intervals of 10 seconds are calibrated to ensure effective call clearing. The relationship between mobile originating calls (MOC) and mobile terminating calls (MTC) is 1:1.

7.1 Call set-up time (parameter 1 in clause 6)

7.1.1 Definition

Call set-up time is the period of time elapsing from the sending of a complete destination address (i.e., target telephone number) to the setting up of a call to the receiving terminal.

7.1.2 Explanation on parameter definition

Call flow diagrams in Figures 7-1, 7-2 and 7-3 for MOC capture two measurement points, "A1" and "A2" for global systems for mobile communications (GSM), code division multiple access (CDMA) and universal mobile telecommunications system (UMTS) respectively.

- "A1" is the time when the mobile station (MS) sends a channel request message.
- "A2" is the time when the MS receives the alerting message from the mobile switching centre (MSC).

The mean time interval between the sending of the channel request message from the calling mobile station (MS) and the calling MS's reception of the alerting message sent from the MSC is the call set-up time; herein mathematically given as A2 minus A1.

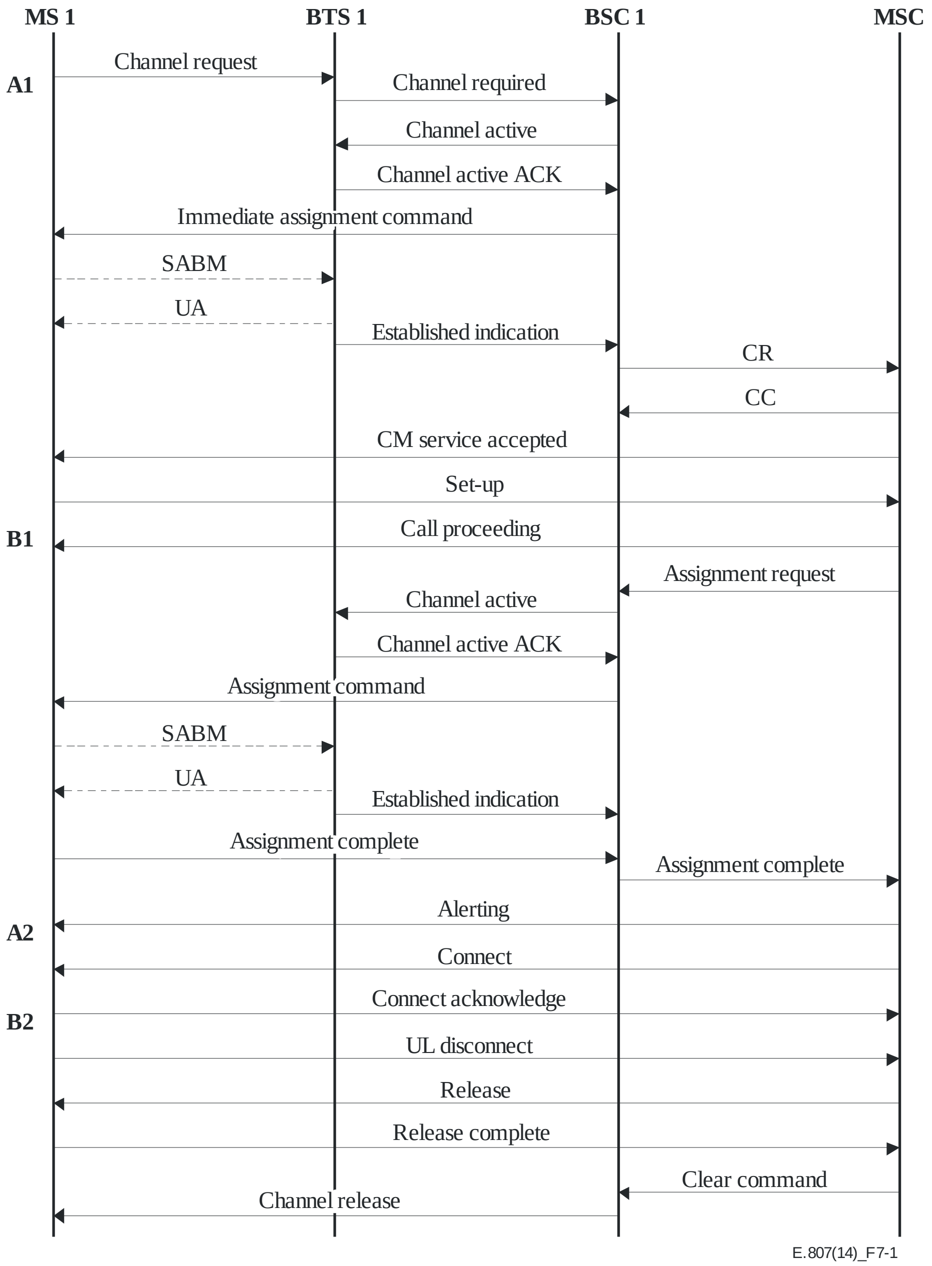

Figure 7-1 – Measurement points in MOC call set-up procedure in GSM

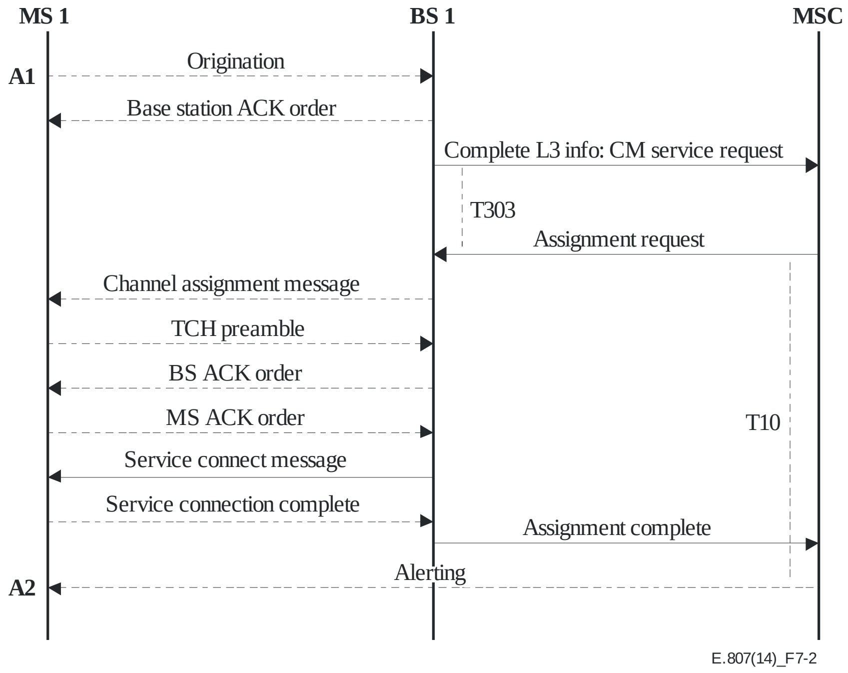

Figure 7-2 – Measurement points in MOC call set-up procedure in CDMA

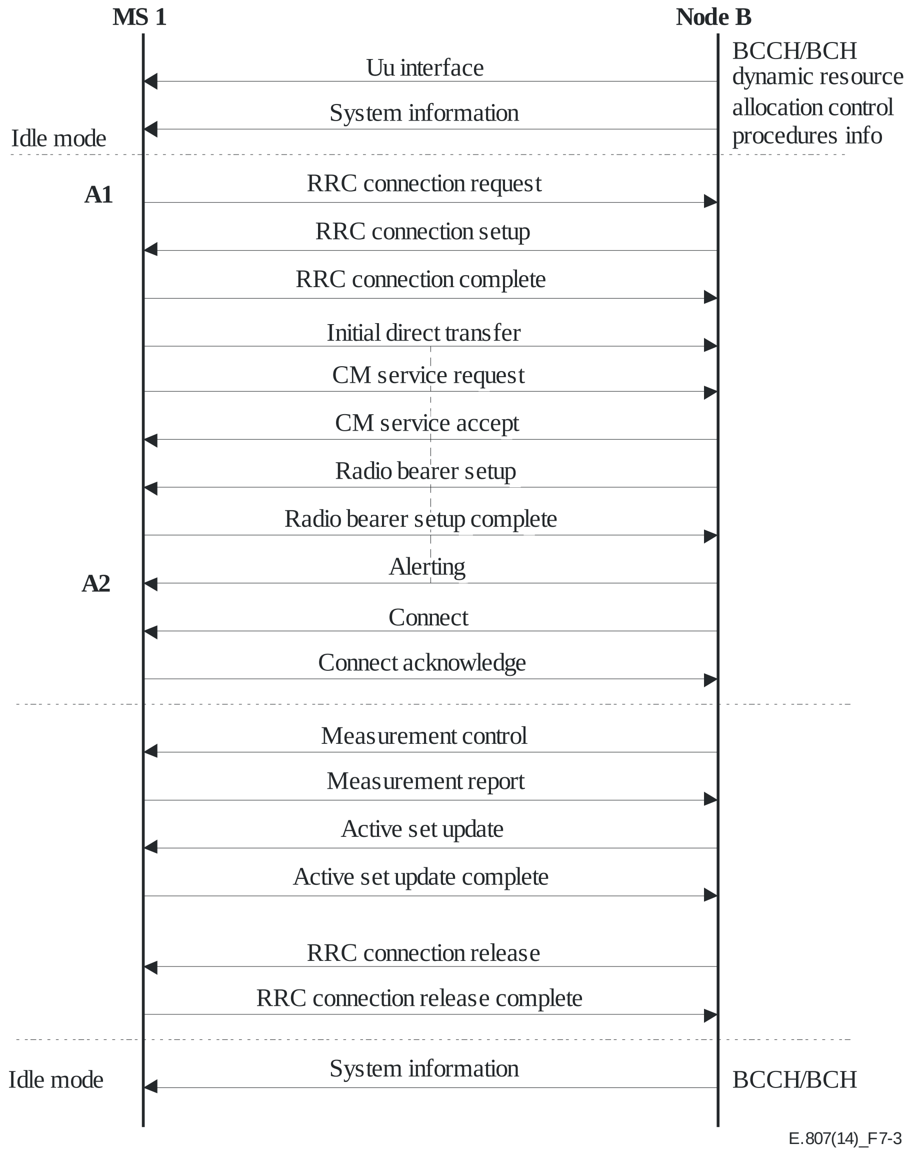

Figure 7-3 – Measurement points in MOC call set-up procedure in UMTS

7.1.3 Equation

$$\text{Call set-up time } [\text{s}] = t_{alerting-signal} - t_{address-sending}$$

7.1.4 Measure

Seconds.

7.1.5 Target

Ninety-five per cent of these connected calls should have a call set-up time less than 10 seconds. Thus, the call set-up time is determined at the 95th percentile of the calls connected.

7.2 Stand-alone dedicated control channel congestion rate (parameter 2 in clause 6)

7.2.1 Definition

Stand-alone dedicated control channel (SDCCH) congestion rate is defined as the probability of failure of accessing a stand-alone dedicated control channel during call set-up.

7.2.2 Explanation on parameter definition

A set-up failed call is triggered by the following two scenarios:

1. No Layer 3 <<CC: Setup>> within 30 s
2. OR a Layer 3<<CC: Channel Release>>

7.2.3 Equation

$$\text{SDCCH Congestion}[\%] = \frac{\text{Number of connect fails due to Immediate Assignment Failures}}{\text{MOC call attempts}} \times 100\%$$

[GSM]

$$\text{RRC Congestion Rate }[\%] = \frac{\text{Number of connect fails due to RRC Connection Setup Failures}}{\text{MOC call attempts}} \times 100\%$$

[UMTS]

7.2.4 Measure

Percentage.

7.2.5 Target

Set-up failed calls due to immediate assignment or radio resource control (RRC) connection set-up failures should not be more than 1 per cent of call attempts.

7.3 Traffic channel congestion rate (parameter 3 in clause 6)

7.3.1 Definition

Traffic channel congestion rate is defined as the probability of failure of accessing traffic channel(s) during call connections.

7.3.2 Explanation on parameter definition

A connect failed call is triggered by the following four scenarios:

1. No Layer 3 <<CC: Connect Acknowledge>> within 30 s after sending a Layer 3 <<CC: Setup>>
2. OR a Layer 3 <<CC: Release>> [GSM]
3. OR a Layer 3 <<CC: RRC Connection Release>> [UMTS]
4. OR a Layer 3<<CC:Channel Release>>

7.3.3 Equation

$$\text{Traffic Channel Congestion }[\%] = \frac{\text{Connect fail due to TCH assignment failures}}{\text{Total number of MOC call attempts}} \times 100\%$$

[GSM]

$$\text{Voice Service Non} - \text{Accessibility Ratio} = \frac{\text{Connect fail due to RAB Setup failures}}{\text{Total number of MOC call attempts}}$$

[UMTS]

7.3.4 Measure

Percentage or ratio.

7.3.5 Target

Connect failed calls due to TCH assignment or radio access bearer (RAB) set-up failures should not be more than 1 per cent of call attempts.

7.4 Call drop rate (parameter 4 in clause 6)

7.4.1 Definition

Voice call drop rate is the probability of a call terminating without the user's action.

7.4.2 Explanation on parameter definition

A drop call indicated by the Break marker is triggered by the following three scenarios in the case of both GSM and UMTS networks:

After the Layer 3 message <<CC: Acknowledge>> is achieved for a call attempt,

1. Phone goes into Idle Mode (Layer 3<<RR: System Information 3>>
2. OR Layer 3 <<Idle Report>>
3. OR Layer 3<<CC:DL Disconnect>>

7.4.3 Equations

$$\text{Drop Rate [\%]} = \frac{\text{Number of calls terminated unwillingly}}{\text{Total number of MOC call attempts}} \times 100\%$$

OR

$$\text{Drop Rate [\%]} = \frac{\text{Number of calls terminated unwillingly}}{\text{Total number of Successful MOC call attempts}} \times 100\%$$

For 'Regulatory' analysis purposes.

$$\text{Drop Rate [\%]} = \frac{\text{Number of calls terminated unwillingly}}{\text{Total number of Successful MOC call attempts}} \times 100\%$$

For 'Network Operator' analysis purposes.

7.4.4 Measure

Percentage or ratio.

7.4.5 Target

Call drop rate should be equal or less than three per cent (3%).

7.5 Call completion rate (parameter 5 in clause 6)

7.5.1 Definition

Call completion rate is defined as the probability that a call, after being successfully set up, has to be maintained during a period of time, ending normally, i.e., according to the user's expectation.

7.5.2 Explanation on parameter definition

Only call attempts that had Layer 3 message <<CC: UL Disconnect>> or <<RRC Connection Release Complete>> are considered.

Completed calls also include "Release failed" calls since the trigger point in the signalling flow of the GSM network is the reception of the <<CC: UL Disconnect>> message by the MS.

7.5.3 Equation

$$\text{Call Completion Rate [\%]} = \frac{\text{Number of normally ended calls}}{\text{Total number of MOC call attempts}} \times 100\%$$

7.5.4 Measure

Percentage or ratio.

7.5.5 Target

Not less than 70 per cent of the call attempts should achieve Layer 3 <<CC: UL Disconnect>> message.

8 Signalling procedures in call origination and termination

When a mobile user, *"calling party"* presses the "Dial" button, a channel request message is sent to the base transceiver station (BTS). The required channel, stand-alone dedicated channel is activated and signalled for assignment to the MS. SDCCH resource unavailability may lead to immediate assignment failure; a phenomenon referred to as SDCCH congestion in the case of GSM mobile systems.

Should an SDCCH be successfully assigned to support authentication and ciphering of the MS, a traffic channel assignment request is then sent by the MSC to the base station controller (BSC). A traffic channel (TCH) assignment command is sent by the BSC and successful seizure is acknowledged by the MS, the "*calling party*" for that matter. Assignment failure in an attempt to originate the call is termed TCH congestion. A connect failed call in GSM is triggered by the following three scenarios:

1. No Layer 3 <<CC: Connect Acknowledge>> within 30 s after sending a Layer 3 <<CC: Setup>>
2. OR a Layer 3 <<CC: Release>>
3. OR a Layer 3<<CC: Channel Release>>

Having paged the *"called"* MS a TCH Assignment request is sent by the MSC to the BSC. Successful TCH seizure triggers an alerting or ringing tone. The mean time interval between the sending of the Channel Request message from the calling MS and the calling MS's reception of the Alerting message sent from the MSC is the call set-up time.

A sent CONNECT message is acknowledged by the *"called"* MS signifying complete call establishment. Untimely Channel Release due to a signalling failure on the part of the network leads to a dropped call otherwise the call is set be "Completed".

8.1 Network factors influencing SDCCH congestion

- **Congestion caused by insufficient signalling resources –** The heavy traffic and burst traffic cause the SDCCH congestion. Proper setting of the number of SDCCHs and TCHs, and the SDCCH dynamic conversion function can relieve the congestion.
- **Congestion caused by improper data configuration –** The SDCCH congestion relates to the relevant parameters of the BSC such as SDCCH availability, LAC and T3101 (the timer used in the immediate assignment procedure), and T3212 (the timer used for periodic updating). If these parameters are set correctly, the SDCCH congestion can be relieved. In addition, if the assignment procedure is set to Late Assignment, the time of the SDCCH being occupied increases, which may lead to congestion.
- **Congestion caused by interference –** Interference on the Um interface in the case of GSM also causes congestion. For example, if the main BCCH in the serving cell and the TCH in the neighbouring cell share the same transceiver (TRX) frequency and BTS base station identity code (BSIC), the handover access on this TCH may be mistaken as random access. As a result, the SDCCH is abnormally allocated and congestion occurs. The excessive receive sensibility can also make the interference signal mistaken as access signal, which leads to congestion.

8.2 Network factors influencing delay in call set-up time

- **Procedure configuration** – The set-up of either an MOC or an MTC involves many procedures such as authentication and ciphering mode setting and relates to multiple NEs such as the MSC, BSC, BTS, and MS. Therefore, the configuration of the call procedures directly determines the length of the call set-up time.
- **Parameter settings** – Incorrect setting of Imm_Ass Retransmit Parameter and Pre-paging function, which is set on the MSC.
- **Routing** – The network equipment of different manufacturers varies. Therefore, a high delay in call set-up time is likely to occur in the case of interworking between equipment of different manufacturers.
- **Hardware, transmission, coverage and interference** – Hardware faults, transmission, coverage, or interference may result in an increase in the call set-up time.

8.3 Network factors influencing TCH congestion

- **Poor coverage –** Affects ability of MS to camp on a BS.
- **Faults occurred during equipment installation, transmission, or on the hardware –** Affect base station (BS) radio resource allocation. The traffic absorption is unbalanced in cells of an area because of faults occurrence during equipment installation, transmission or on the hardware or clock. As a result, TCHs in certain cells are overloaded whilst in some cells are they are idle.
- **Network interference –** Affects BS radio resource allocation.
- **Incorrect parameter settings** – Traffic balancing, handover, and Rx_Min_Access_level parameter when poorly set can lead to high traffic and consequently failed calls due to TCH busy.
- **Software version problems –** When the BSC is upgraded or a new version is used in the existing network, the software of earlier versions may be incompatible with the new platform (for example, BSC6000) or software bugs may increase the TCH congestion rate. Therefore, we should locate and rectify the fault caused by software problems by taking proper measures.

8.4 Network factors influencing dropped calls

According to user complaints and network optimization experience, the major factors that affect the TCH call drop rate are as follows:

- hardware failure and transmission alarms (i.e., fibre cuts, E 1 errors and loss of frames)
- improper parameter setting
- network interference
- poor coverage
- imbalance between uplink and downlink.

9 Conclusion

This Recommendation provides step-by-step signalling procedures required when originating and terminating a voice call; more so indicating important Layer 3 reference points used in analysing the five user-centric parameters afore-discussed. A relationship has been established between network performance and voice service quality.

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