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OF ITU

Q.764

Amendment 1
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SERIES Q: SWITCHING AND SIGNALLING
Specifications of Signalling System No. 7 – ISDN user part

Signalling system No. 7 – ISDN user part signalling procedures

Amendment 1

ITU-T Recommendation Q.764 – Amendment 1

(Formerly CCITT Recommendation)

ITU-T Q-SERIES RECOMMENDATIONS

SWITCHING AND SIGNALLING

SIGNALLING IN THE INTERNATIONAL MANUAL SERVICE	Q.1–Q.3
INTERNATIONAL AUTOMATIC AND SEMI-AUTOMATIC WORKING	Q.4–Q.59
FUNCTIONS AND INFORMATION FLOWS FOR SERVICES IN THE ISDN	Q.60–Q.99
CLAUSES APPLICABLE TO ITU-T STANDARD SYSTEMS	Q.100–Q.119
SPECIFICATIONS OF SIGNALLING SYSTEMS No. 4 AND No. 5	Q.120–Q.249
SPECIFICATIONS OF SIGNALLING SYSTEM No. 6	Q.250–Q.309
SPECIFICATIONS OF SIGNALLING SYSTEM R1	Q.310–Q.399
SPECIFICATIONS OF SIGNALLING SYSTEM R2	Q.400–Q.499
DIGITAL EXCHANGES	Q.500–Q.599
INTERWORKING OF SIGNALLING SYSTEMS	Q.600–Q.699
SPECIFICATIONS OF SIGNALLING SYSTEM No. 7	Q.700–Q.799
General	Q.700
Message transfer part (MTP)	Q.701–Q.709
Signalling connection control part (SCCP)	Q.711–Q.719
Telephone user part (TUP)	Q.720–Q.729
ISDN supplementary services	Q.730–Q.739
Data user part	Q.740–Q.749
Signalling System No. 7 management	Q.750–Q.759
ISDN user part	**Q.760–Q.769**
Transaction capabilities application part	Q.770–Q.779
Test specification	Q.780–Q.799
Q3 INTERFACE	Q.800–Q.849
DIGITAL SUBSCRIBER SIGNALLING SYSTEM No. 1	Q.850–Q.999
PUBLIC LAND MOBILE NETWORK	Q.1000–Q.1099
INTERWORKING WITH SATELLITE MOBILE SYSTEMS	Q.1100–Q.1199
INTELLIGENT NETWORK	Q.1200–Q.1699
SIGNALLING REQUIREMENTS AND PROTOCOLS FOR IMT-2000	Q.1700–Q.1799
SPECIFICATIONS OF SIGNALLING RELATED TO BEARER INDEPENDENT CALL CONTROL (BICC)	Q.1900–Q.1999
BROADBAND ISDN	Q.2000–Q.2999

For further details, please refer to the list of ITU-T Recommendations.

ITU-T Recommendation Q.764

Signalling system No. 7 – ISDN user part signalling procedures

AMENDMENT 1

Summary

This amendment contains clarifications and modifications to ITU-T Q.764 (1999) in order to accommodate the requirements of DME signalling, Global Call Reference, Inter-nodal Traffic Group Identification, and Carrier Selection Indication.

Source

Amendment 1 to ITU-T Recommendation Q.764 was prepared by ITU-T Study Group 11 (2001-2004) and approved under the WTSA Resolution 1 procedure on 13 July 2001.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

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, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

ITU-T Recommendation Q.764

Signalling system No. 7 – ISDN user part signalling procedures

AMENDMENT 1

1) Clause 1.2, References

Modify reference [19] as follows:

[19] ITU-T Q.1902.3 (2001)~~Q.763 (1999)~~, *Bearer independent call control (CS2) protocol and s~~S~~ignalling s~~S~~ystem No. 7 – ISDN u~~U~~ser p~~P~~art formats and codes*.

2) Clause 2.14, MTP pause/resume

Modify the last paragraph as follows:

On the reception of a MTP resume primitive, the ISDN User Part takes the following action:

– If the affected destination is not a destination (Signalling Point) known by the ISDN User Part (not connected by circuits to the exchange) no action takes place.

– If the affected destination is a destination (Signalling Point) known by the ISDN User Part, the circuits in the idle state can be used for calls immediately; or, as a national option the circuits in the idle state remain locally blocked, and a non-call control message requiring a response shall be sent to the distant ISUP. On receipt of the response message (or any other signalling message) from the distant ISUP, the local blocking resulting from the previously received MTP pause primitive shall be removed.

  Normal call release procedures that may have started during the period of signalling isolation continue and as such will ensure that affected circuits are returned to the idle state.

3) New clauses 2.21 to 2.25

Add the following:

2.21 Signalling procedure for tandem mode operation of Digital Multiplexing Equipment with Low-bit-rate Voice CODEC (DME with LVC)

2.21.1 Introduction

In order that Digital Multiplexing Equipment with Low-bit-rate Voice CODEC (DME with LVC) can operate in tandem mode without unnecessary decompression and recompression of voice, by means of:

1) routing of calls to appropriate circuits with/without DMEs with LVC; and

2) enabling/disabling of voice decompression/recompression in DMEs with LVC,

the tandem mode operation signalling procedures are used on a per-call basis to convey information between the exchanges about the voice compression type(s) used in the preceding network for the call, whether voice is compressed between the exchanges and the voice compression status (if voice is compressed).

2.21.2 Signalling procedures

2.21.2.1 Actions at originating exchange

When the originating exchange receives a call from the calling user, based on the information of the received call control message and/or the system configuration of the exchange, the exchange analyzes whether the outgoing circuit including DME with LVC can be selected.

If the originating exchange judges that the outgoing circuit including DME with LVC can not be selected, the originating exchange routes the call to the outgoing circuit without DME with LVC.

If the originating exchange judges that the outgoing circuit including DME with LVC can be selected, the exchange:

- routes the call to the outgoing circuit including DME with LVC;
- enables compression of voice in the DME with LVC over the outgoing circuit;
- adds Type of voice compression and Compression status of Coding Decoding Processing Parameter (CDP-P) in IAM message which are set as the used voice compression type and "Compressed" regarding the used voice compression type; and
- sets the used voice compression type into User information layer 1 protocol of User Service Information Parameter (USI-P) in IAM message regarding the used voice compression type.

2.21.2.2 Actions at intermediate exchange

When the intermediate exchange receives a call, based on the information of the received call control message and/or the system configuration of the exchange, the intermediate exchange judges the configuration case of DME with LVC from the following four cases:

- *Case 1*

  The DMEs with LVC with the same type of voice compression capability are used over the incoming circuit and the outgoing circuit.
- *Case 2*

  The DMEs with LVC is included in the circuit on the incoming side and the DME with LVC is not included in the circuit on the outgoing side.
- *Case 3*

  The DME with LVC is included in the circuit on the outgoing side and the DME with LVC is not included in the circuit on the incoming side.
- *Case 4*

  The DME with LVC with the different types of voice compression capability are used over the incoming circuit and the outgoing circuit.

2.21.2.2.1 Actions at intermediate exchange for Case 1

The intermediate exchange:

- disables decompression of voice in the DME with LVC over the incoming circuit and recompression of voice in the DME with LVC over the outgoing circuit;
- sets the used voice compression type and "Compressed" into Type of voice compression and Compression status of Coding Decoding Processing Parameter (CDP-P) in IAM message which are same as those of the received call control message; and
- sets the used voice compression type into User information layer 1 protocol of User Service Information Parameter (USI-P) in IAM message which are same as those of the received call control message.

2.21.2.2.2 Actions at intermediate exchange for Case 2

The intermediate exchange:

– enables decompression of voice in DME with LVC over the incoming circuit;

– sets "Decompressed" into Compression status of Coding Decoding Processing Parameter (CDP-P) in IAM message regarding the used voice compression type; and

– sets "μ-law/A-law" into User information layer 1 protocol of User Service Information Parameter (USI-P) in IAM message.

2.21.2.2.3 Actions at intermediate exchange for Case 3

When the intermediate exchange receives a call, based on the received call control message, the intermediate exchange analyzes whether the new voice compression causes severe voice quality degradation.

If the intermediate exchange judges that severe voice quality degradation is caused, the intermediate exchange routes the call to the outgoing circuit without DME with LVC.

If the intermediate exchange judges that severe voice quality degradation is not caused, the intermediate exchange:

– enables compression of voice in the DME with LVC over the outgoing circuit;

– adds Type of voice compression and Compression status of Coding Decoding Processing Parameter (CDP-P) in IAM message which are set as the used voice compression type and "Compressed" regarding the used voice compression type; and

– sets the used voice compression type into User information layer 1 protocol of User Service Information Parameter (USI-P) in IAM message regarding the used voice compression type.

2.21.2.2.4 Actions at intermediate exchange for Case 4

When the intermediate exchange receives a call, based on the received call control message, the intermediate exchange analyzes whether the new voice compression causes severe voice quality degradation.

If the intermediate exchange judges that severe voice quality degradation is caused, the intermediate exchange routes the call to the outgoing circuit without DME with LVC.

If the intermediate exchange judges that severe voice quality degradation is not caused, the intermediate exchange:

– enables decompression of voice in the DME with LVC over the incoming circuit and compression of voice in the DME with LVC over the outgoing circuit;

– sets "Decompressed" into Compression status of Coding Decoding Processing Parameter (CDP-P) in IAM message regarding the used voice compression type;

– adds Type of voice compression and Compression status of Coding Decoding Processing Parameter (CDP-P) in IAM message which are set as the used voice compression type and "Compressed" regarding the used voice compression type; and

– sets the used voice compression type into User information layer 1 protocol of User Service Information Parameter (USI-P) in IAM message regarding the used voice compression type.

2.21.2.3 Actions at destination exchange

When the destination exchange receives, based on the information of the received call control message, the destination exchange analyzes whether the call is received via the incoming circuit including DME with LVC.

If the destination exchange judges that the call is received via the incoming circuit including DME with, the destination exchange:

– enables decompression of voice in DME with LVC over the incoming circuit.

It is for further study whether the information included in Coding Decoding Processing Parameter (CDP-P) is sent to the called party in the case of a subscriber with digital access.

2.22 Handling of national use elements at an international gateway exchange

Unless a bilateral or multilateral agreement is reached among the network operators concerned, messages, parameters and parameter values marked as "national use" are not valid in the international network. Thus, an outgoing or incoming international gateway shall ensure that any "national use" messages/parameters/values received from its associated national network are not passed on.

NOTE – This requirement can be satisfied, for example, by the international gateway:

– fully implementing the appropriate national procedures, and any necessary interworking functions; or

– treating all "national use" codespace as unrecognized, and providing "pass-on not possible" treatment.

2.23 Inter-nodal traffic group identification

The Inter-nodal Traffic Group Identifier parameter may be included in the IAM in order to enable classification of calls between adjacent nodes. It identifies the logical traffic group to which the call belongs; i.e. this identifier is of significance only between two adjacent nodes. These classifications could, for example, be used to make a distinction between different service sets. These classifications are not standardized.

2.23.1 Sending Inter-nodal traffic group identification

If needed for the chosen outgoing route, the exchange shall include the Inter-nodal Traffic Group Identifier parameter, populated according to the relevant classification. This classification may depend on a classification received on the incoming side.

NOTE – The Parameter Compatibility Instruction Indicators for this parameter should be set to ensure that the parameter is not passed on at a node that does not recognize the parameter.

2.23.2 Receiving Inter-nodal traffic group identification

The traffic group identification received in a Inter-nodal Traffic Group Identifier parameter is used according to the relevant classification. A received Traffic Group Identifier parameter may be used to influence the routing of the call.

2.24 Carrier selection information (national use)

2.24.1 Action required at the originating exchange

If a Carrier Selection is invoked by the user (reception of carrier selection information from the access) or by the network operator, the exchange shall send the Carrier Selection Information (CSI) parameter in the IAM.

NOTE – The carrier selection information received from the access can be provided by a short prefix conveyed in the called party number or by other means, depending on the access signalling system.

The CSI parameter shall be set as follows:

• If call-per-call Carrier Selection is not invoked and there is a preselected carrier, then the CSI parameter is set to *"selected carrier identification pre-subscribed, and no input by calling party"* (value 1).

• If a carrier is call-per-call selected, then the CSI parameter is set to *"Carrier selected by input of calling party"* (value 10) (see Note below).

- If a carrier is selected by the network operator to which belongs the exchange, then the CSI parameter is set to *"carrier selected by a network operator"* (value 11).

If no Carrier Selection is invoked, the CSI parameter shall not be sent.

NOTE – A coding giving more accurate information could possibly be used (*"Selected Carrier identification pre-subscribed and input by calling party"* (value 2) or *"Selected Carrier identification not pre-subscribed, and input by calling party"* (value 4)). The reason for using a generic coding (value 10) comes from regulation in some countries which protects privacy of the calling party.

2.24.2 Action required at an intermediate exchange within the originating network

The intermediate exchange shall pass unchanged the CSI parameter to the subsequent exchange.

2.24.3 Action required at an outgoing national gateway exchange

The outgoing national gateway exchange will pass on the parameter transparently.

2.24.4 Action required at an incoming national gateway exchange

a) In case the network to which belongs the gateway exchange is explicitly selected:

   The handling of the content of the CSI parameter is a network matter. However, the parameter shall not be sent to any subsequent network.

b) In case the network to which belongs the gateway exchange is not explicitly selected:

   The call is routed through the network with the CSI parameter unchanged.

2.24.5 Action required at the destination exchange

No special action required.

2.24.6 Action required at an international gateway exchange

The international gateway exchange shall discard the CSI parameter.

2.25 Global Call Reference

The Global Call Reference parameter is generated by the first exchange in a call path that requires a globally unique call reference to be associated with a particular call.

The Global Call Reference is a combination of a Network ID field, a Node ID field and a Call Reference ID field. The Network ID field will uniquely identify the network; the Node ID field will uniquely identify the node within this network that generates the Global Call Reference parameter. The Call Reference ID field will be a unique number generated on a per-call instance within this node.

The Global Call Reference parameter is sent in the forward direction in the IAM.

The intermediate exchange shall pass this parameter unchanged.

The Global Call Reference parameter shall be stored in the nodes which require this reference according to the needs of the application that uses the information.

NOTE 1 – The Global Call Reference parameter may typically be used for off-line purposes (e.g. to be stored for billing applications).

NOTE 2 – A exchange may delete a received Global Call Reference parameter (e.g. at an outgoing gateway exchange).

NOTE 3 – A received Global Call Reference may be overridden (e.g. at an incoming gateway exchange).

SERIES OF ITU-T RECOMMENDATIONS

Series A	Organization of the work of ITU-T
Series B	Means of expression: definitions, symbols, classification
Series C	General telecommunication statistics
Series D	General tariff principles
Series E	Overall network operation, telephone service, service operation and human factors
Series F	Non-telephone telecommunication services
Series G	Transmission systems and media, digital systems and networks
Series H	Audiovisual and multimedia systems
Series I	Integrated services digital network
Series J	Cable networks and transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Construction, installation and protection of cables and other elements of outside plant
Series M	TMN and network maintenance: international transmission systems, telephone circuits, telegraphy, facsimile and leased circuits
Series N	Maintenance: international sound programme and television transmission circuits
Series O	Specifications of measuring equipment
Series P	Telephone transmission quality, telephone installations, local line networks
Series Q	**Switching and signalling**
Series R	Telegraph transmission
Series S	Telegraph services terminal equipment
Series T	Terminals for telematic services
Series U	Telegraph switching
Series V	Data communication over the telephone network
Series X	Data networks and open system communications
Series Y	Global information infrastructure and Internet protocol aspects
Series Z	Languages and general software aspects for telecommunication systems

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