ETSI TS 148 103 V13.0.0 (2016-01)

TECHNICAL SPECIFICATION

Digital cellular telecommunications system (Phase 2+);
Base Station System - Media GateWay (BSS-MGW) interface;
User plane transport mechanism
(3GPP TS 48.103 version 13.0.0 Release 13)

A GLOBAL INITIATIVE

GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

Reference

RTS/TSGG-0248103vd00

Keywords

GSM

ETSI

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Contents

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1 Scope

The present document specifies the User Plane data transport protocols used between BSSs and the Core Network (MGWs) across the A interface. The main purpose of the present document is the AoIP description, however for the sake of completeness the AoTDM case is described as well.

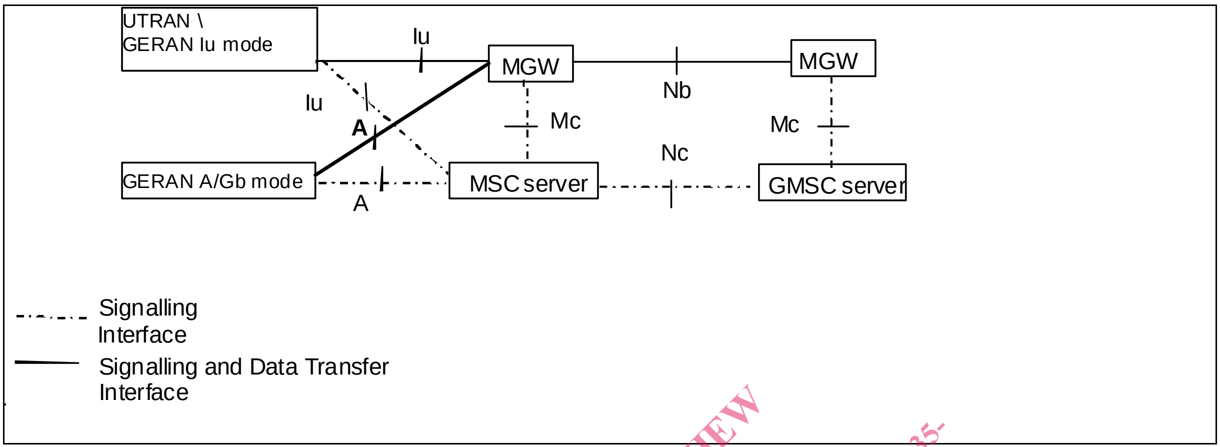

Figure 1.1: CS core network logical architecture

Note that the present document does not preclude any Core Network Session Control Protocol implementation (BICC or SIP-I).

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
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[1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[2] IETF RFC 791: "Internet Protocol (IP)".

[3] IETF RFC 2460: "Internet Protocol, Version 6 (IPv6)".

[4] IETF RFC 768: "User Datagram Protocol. (UDP)".

[5] IETF RFC 3550: "RTP: A Transport Protocol for Real Time Applications".

[6] 3GPP TS 29.414: "Core network Nb Interface data transport and transport signalling".

[7] IETF RFC 3551: "RTP Profile for Audio and Video Conference with Minimal Control".

[8] 3GPP TR 29.814: "Feasibility Study on Bandwidth Savings at Nb Interface with IP transport".

[9] IETF RFC 4040: "RTP Payload Format for a 64 kbits/s Transparent Call"

[10] IETF RFC 4867: "RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs"

[11] IETF RFC 2198: "RTP Payload for redundant Audio Data"

[12] 3GPP TR 43.903 A-Interface over IP Study (AINTIP)

[13] 3GPP TS 48.001 "Base Station System – Mobile-services Switching Centre (BSS - MSC) interface; General aspects"

[14] ITU-T Recommendation G.705: "Characteristics of plesiochronous digital hierarchy (PDH) equipment functional blocks".

[15] ANSI T1.102-1993: "Digital Hierarchy - Electrical Interface".

[16] ITU-T Recommendation G.711: "Pulse Code Modulation (PCM) of voice frequencies".

[17] 3GPP TS 48.020: "Rate adaption on the Base Station System - Mobile-services Switching Centre (BSS - MSC) interface".

[18] IETF RFC 5993 (2010) "RTP Payload Format for Global System for Mobile Communications Half Rate (GSM-HR)".

[19] 3GPP TS 48.008: "Mobile Switching Centre - Base Station System (MSC-BSS) interface; Layer 3 specification".

[20] 3GPP TS 26.102: "Adaptive Multi-Rate (AMR) speech codec; Interface to Iu, Uu and Nb".

[21] 3GPP TS 23.284: "Local Call Local Switch; Stage 2".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in TR 21.905 [1] and the following apply.

Intra-BSS call: A mobile to mobile voice call involving two mobile stations connected to the same BSS.

Local call: An Intra-BSS call that can be locally switched by the BSS. For details on the Local Switch Service see 3GPP TS 23.284 [21].

Access MGW: The MGW interfacing to the Radio Access Network

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in TR 21.905 [1] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in TR 21.905 [1].

AoIP	A (interface) over IP
AoTDM	A (interface) over TDM
APP	APPlication
BICC	Bearer Independent Call Control
BSS	Base Station Subsystem
CS	Circuit-Switched
CSData	CS Data and fax
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ITU-T	International Telecommunications Union-Telecommunication sector
MGW	Media GateWay
PCM	Pulse-Coded Modulation
RFC	Request For Comment

RTP	Real-Time Transport Protocol
RTCP	Real-Time Transport Control Protocol
SIP-I	Session Initiation Protocol with encapsulated ISUP
SSRC	Synchronisation Source
SVC	Switched Virtual Circuit
TDM	Time-Division Multiplexing
UDP	User Datagram Protocol
UP	User Plane

4 Transport over TDM

4.1 General

The present chapter describes the transport on the A Interface UP over E1/T1 interface; for more information see 3GPP TS 48.001 [13]. Figure 4.1 shows the protocol stack for the transport network user plane on the A interface.

Payload (PCM encoded speech or CSData)
E1 or T1

Figure 4.1: TDM Protocol stack for the A interface user plane

Layer 1 shall utilise digital transmission:

- at a rate of 2 048 kbit/s with a frame structure of 32 x 64kbit/s time slots, as specified in ITU-T Recommendation G.705 [14] clause 3 for E1 interface; or
- at a rate of 1 544 kbit/s with a frame structure of 24 x 64 kbit/s time slots, as specified in T1.102 specification for T1 interface [15].

The payload is either PCM encoded speech (see ITU-T Recommendation G.711 [16]) or CSData (see 3GPP TS 48.020 [17]).

4.2 Transport during local switching

When a local switch path is established in the BSS for a local call, the user plane between the BSS and the Access MGWs for both the call legs in uplink and downlink shall not be released.

5 Transport over IP

5.1 General

The present chapter describes the transport of the A-Interface User Plane Payload by RTP/UDP/IP. Figure 5.1 shows the protocol stack.

Payload (PCM-coded speech, or compressed speech or CSData)
RTP
UDP
IPv4 or IPv6

Figure 5.1: IP Protocol stack for the A-Interface user plane

The specific carrying way at physical/link layer of the IP protocol is not limited by the present document; if Ethernet is adopted, link layer will be MAC protocol while if IPoE1 or POS is adopted, link layer will be PPP protocol. Nevertheless at least Ethernet should be supported.

5.2 IP

IPv4 (RFC 791 [2]) shall be supported

IPv6 (RFC 2460 [3]) may be supported as an option.

One BSS/MGW pair may be connected via several IP interfaces.

5.3 UDP

The UDP Protocol (see RFC 768 [4]) shall be applied.

Two consecutive port numbers shall be used at each BSS/MGW for the RTP connection and for the optional RTCP connection that corresponds to a single A interface UP connection. Two such consecutive port numbers are termed "port number block" in what follows. The first port number shall be even and shall be assigned to the RTP protocol. For a given BSS/MGW, the same port shall be used to send and to receive RTP PDUs. The next port number shall be assigned to the RTCP protocol. This port shall be reserved even if the optional RTCP protocol is not used.

If multiplexing is applied with or without header compression, the source UDP port number shall indicate the local termination used to combine the multiplexed packet and the destination UDP port number shall indicate the remote port number where PDUs are demultiplexed. The negotiation of whether multiplexing is applied and of the destination UDP port is described in sub-clause 5.5.3.2. If multiplexing was negotiated for an A interface UP user plane connection, the BSS/MGW may apply multiplexing by sending all packets of the user plane connection (multiplexed with other user plane connections packets) towards the negotiated destination UDP port.

5.4 Transport without RTP multiplexing

5.4.1 Introduction

User Plane transport without RTP multiplexing is default and shall be supported. It shall be applied after call setup, until a User Plane transport with RTP-multiplexing is negotiated via RTCP, see clause 5.5. RTCP, see RFC 3550 [5] may be applied in AoIP, it is optional. A BSS or MSC may ignore incoming RTCP packets on AoIP.

5.4.2 RTP

The RTP protocol (see RFC 3550 [5]) shall be applied.

5.4.2.1 RTP Header

The RTP Header Fields shall be used as described in the following sub-clauses:

5.4.2.1.1 Version

RTP Version 2 shall be used.

5.4.2.1.2 Padding

Padding shall not be used.

5.4.2.1.3 Extension

The RTP Header extension shall not be used.

5.4.2.1.4 Contributing Source (CSRC) count

There is zero CSRC.

5.4.2.1.5 Marker Bit

The marker bit shall be used as specified for the RTP profile applicable for the used payload. If AMR or AMR-WB speech is received via the GSM radio interface, then an ONSET frame precedes the first speech frame. This ONSET Frame is transported in a separate RTP packet and shall also have the Marker Bit set to "1". Also the next RTP packet, which contains the first speech frame of the talkspurt, shall have the Marker Bit set to "1". In case of lost speech frames due to radio errors or due to FACCH frame stealing the Marker bit shall not be set in the first speech frame after that interruption.

5.4.2.1.6 Payload Type

See sub-clause 5.4.2.2.

5.4.2.1.7 Sequence Number

The sequence number shall be supplied by the source (BSS or MGW) of an RTP packet. RTP sequence numbering is based on sent RTP packets, not on expected speech frames. If frames are lost or stolen on the radio interface and the codec does not support Bad or No_Data frame types, no RTP packet is sent in uplink direction and the RTP Sequence Number is not incremented, until a next frame is sent in RTP. This ensures that the receiver of the RTP stream can detect lost RTP packets by inspecting the RTP Sequence Number.

5.4.2.1.8 Timestamp

The timestamp shall be supplied by the source (BSS or MGW) of a RTP PDU. Depending of the (pseudo) codec used a clock frequency of either 8 or 16 kHz shall be used, as described in sub-clause 5.4.2.2. In case of lost or stolen frames on the radio interface or in case of an interruption of the RTP stream due to handover, the RTP Timestamp shall be incremented as if no frame would have been lost. This ensures that the receiver of the RTP stream can regenerate the time signal correctly.

NOTE: IETF RFC 3550 [5] specifies that the RTP timestamp is based on the sampling instant of the source Encoder. In case of a circuit switched radio interface the source Encoder for the uplink is within the mobile station. But the radio interface does not support the transfer of a RTP Timestamp. The clock synchronisation between mobile station and BSS is, however, very precise and so the BSS can take the role of the source Encoder and provide the RTP time stamp.

5.4.2.1.9 Synchronisation Source (SSRC)

The source (BSS or MGW) of a RTP PDU shall supply a SSRC. The receiver of a RTP PDU may ignore the SSRC if it does not use RTCP.

5.4.2.1.10 CSRC list

This list is empty, i.e. not present.

5.4.2.2 RTP Payload

The packing of the RTP Payload for each Speech Codec Type is specified in TS 26.102 [20].

When defined as such by IETF for a given Speech Codec (see RFC 3550 [5]) and RFC 3551 [7]) the "Static" Payload Type is used; otherwise for RTP profiles for which IETF defines the Payload Type as "Dynamic" (e.g. for AMR codecs, see RFC4867 [10]) a fixed Payload Type is defined for AoIP in the range of the dynamic Payload Types, see below.

The mapping between transported RTP payloads and their associated Payload Type is as follows: