# User Description, Packet Abis over IP

USER DESCRIPTION

326/1553-HSC 103 12/25 Uen C

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# Contents

# 1 Introduction

The feature Packet Abis over IP uses IP networks for transmission between BSC and BTS, instead of a TDM network. Packet Abis over IP supports IP over Ethernet (and also over E1/T1), but the intermediate transport network may in principle use any available technology.

Compared to TDM networks, Abis over IP provides significantly higher transmission capacity per bandwidth resource, since it uses the same packetized framework as the feature Packet Abis over TDM. The bandwidth is utilized more efficiently by letting signaling, speech and data share the same wideband connection. The transmission capacity is used like a pool of resources. All transmission resources are occupied only when they are actively used. Transport sharing with other services, such as 3G, is also possible when using Abis over IP.

## 1.1 Reader Guide

This document describes the feature Packet Abis over IP to operators and other users of the feature. The feature name Packet Abis over IP was introduced in release G11B. In previous releases the feature name was Abis over IP. In this document as well as in related documents any of the names Packet Abis over IP or Abis over IP may be used.

This User Description is based on the document 326/1553-HSC 103 12/24 Uen, revision C and is valid for GSM RAN release G16B and onwards. New functionality works for stated release and onwards, unless stated otherwise. It focuses on Ethernet based Abis over IP (not E1/T1 based). The aim of the document is to make the initiated reader get a grasp of Abis over IP and be guided in configuring the feature. Since Abis over IP is a very complex feature, the total picture is provided by the help of several documents.

Documents needed for the overall picture of Packet Abis over IP feature:

- *GSM RAN IP RAN System Description* (Reference [1]) briefly describes IP network functionality inside the BSS nodes and integration between the nodes.
- User Description *Packet Abis over TDM* (Reference [2]) describes the packetizing framework which Packet Abis over IP uses. Prior to G11B this feature was called Abis Optimization.
- User Guide *IP over E1/T1* (Reference [3]) describes the possibility to run Abis over IP in an E1/T1 network.
- User Guide *Abis over IP Network Characteristics* (Reference [29]) describes requirements on IP network characteristics, and Abis over IP impacts on end user performance.

Documents supplementing this document in important areas:

- User Description *Baseband Radio Node for GSM* (Reference [30]) describes the handling of Baseband Radio Node.
- User Description *PGW Load Distribution* (Reference [4]) describes the distribution of load between PGW-RPs internally in BSC for the Packet Abis features.
- User Description *CTH Load Distribution* (Reference [5]) describes the distribution of load between PGWs hosted by CTH-RPs in Evo Controller 8200/BSC with RP hardware of type Evo Processor Board (EPB).
- User Description *Channel Allocation Optimization* (Reference [6]) describes functions used by load regulation functions triggered in Abis over IP.
- User Guides *Synchronization* (Reference [7]) and *Manage Network Synchronization* (Reference [32]) describes various radio network synchronization alternatives.
- User Description *Radio Network Statistics* (Reference [8]) describes methods to measure the radio network performance and subscriber perceived quality with a multitude of statistics accessible in BSC.
- *User Description, GSM RAN IP Security* (Reference [9]) describes security aspects of using IP transport in GSM RAN.

Guides for dimensioning and planning:

- *Packet Abis Dimensioning Guideline* (Reference [10]) describes rule of thumbs and concepts that are good to know while dimensioning packet Abis.
- *BSC/TRC and BSC Hardware Dimensioning* (Reference [11]) describes how to equip BSC with hardware to achieve desirable performance.
- User Guide *BSC IP Addressing* (Reference [12]) provides recommendations for the IP address plan in the BSC used by Abis over IP.

Guides for operation and maintenance:

- User Guide *Managing Abis over IP* (Reference [13]) describes the OSS perspective on Abis over IP.
- System Administrator Guide *CDM, Configuration Data Mart* (Reference [34]) and Function Description *CDM, Configuration Data Mart* (Reference [35]) describe OSS-RC management of the Baseband Radio Node.

Related functions and features:

- The packing function described in the User Description *Packet Abis over TDM* (Reference [2]) is recommended for more efficient bandwidth utilization of Abis over IP.

- User Guide *Site Router* (Reference [14]) describes auxiliary site equipment in SIU and TCU and its influence on the common transmission.
- User Guide *Abis over IP, Non Co-located STN and BTS Support* (Reference [15]) describes special dimensioning and configuration issues for Abis over IP when the SIU STN is connected to one or more BTSs not co-located with the STN, but connected over a network.
- User Description *Abis Local Connectivity* (Reference [16]) describes the feature Abis Local Connectivity to reduce the traffic on Abis resulting from local calls within the same STN (only possible for Abis over IP).
- User Guide *RAN Transport Sharing and QoS* (Reference [17]) describes how GSM Abis over IP can share its transport network with WCDMA, LTE and CDMA.
- User Description *Flexible Abis* (Reference [18]) describes an older feature that is superseded by Packet Abis features. The document describes some of the overload prevention features used by Abis over IP.

## 1.2 Main changes in Ericsson GSM RAN G16B

The following new technical solution is introduced in G16B:

- The STN functionality is now also supported and realized on the Baseband T605 HW.
- Introduction of the Baseband Radio Node.

## 1.3 Main changes in Ericsson GSM RAN G15B

The following new technical solution is introduced in G15B:

- Functionality to omit PS-scheduling due to high load introduced.
- Improved Abis Lower Characteristics.

## 1.4 Main changes in Ericsson GSM RAN G14B

The following new technical solution are introduced in G14B:

- Load regulation and traffic shaping are introduced on Abis Lower.

## 1.5 Main changes in Ericsson BSS G13B

The following new technical solution are introduced in G13B, compared to G12B:

- New Regional Processor (RP) application Combined Traffic Handler (CTH) introduced in BSC. The Combined Traffic Handler (CTH) is used in Evo Controller 8200/BSC with RP hardware of type Evo Processor Board (EPB). The PGW is hosted by the CTH for Evo Controller 8200/BSC.

# 2 Capabilities

Upgrade to Abis over IP requires new hardware. In BSC, PGWs are required (may already be in place with the feature Packet Abis over TDM). On the BTS side of Abis, STNs or Baseband Radio Nodes are required. The STNs can be either SIUs, TCUs, Baseband T605 or RBS 2409. For synchronization in IP over Ethernet, a TimeServer is needed. A Security Gateway (SEGW) is needed for the recommended level of safety.

## 2.1 BSC Hardware

Both the PGW and the BSC IP network interfaces are prerequisites for Abis over IP.

The BSC IP network interfaces are realized in the BSC LAN Switches and the BSC Network Interface - Ethernet (NWI-E).

The PGW application is realized in the PGW-RP or as a part of the Combined Traffic Handler RP (CTH-RP). The Regional Processor application CTH was introduced in BSC G13B. The Combined Traffic Handler (CTH) is used in Evo Controller 8200/BSC.

In this document PGW is used both for the PGW-RP application and the PGW part of the CTH-RP application.

The PGW terminates IP on the BSC side, and is the hardware in the BSC used to support Abis over IP. The PGW handles speech, (E)GPRS and signaling in the same RP. The PGW can handle a mix of Packet Abis over IP or TDM transmission, but each TG can only use one type of transmission mode.

The amount of traffic one PGW can handle depends on network configuration and transmission load. For details on PGW dimensioning see Reference [11].

## 2.2 STN and BTS Hardware

On the BTS site the IP network is terminated in the STN for RBS or in the Baseband Radio Node. See the product description in relevant node CPI for more information. For specific information on BTS supporting Abis over IP see the GSM/BTS Software release note of the RBS versions used. Cascaded sites are not supported.

- In RBS 2409 the STN hardware is fully integrated with the BTS hardware. One RBS 2409 supports one TG (see details in Table 1).

- The hardware called SIU is a standalone STN. SIU supports all Macro RBSs. One SIU supports up to six TGs (see details in Table 1). A SIU has 16 E1/T1 that can be used for Abis lower or Abis upper (ML-PPP)

- The hardware called TCU is an STN made to fit into the RBS6000 cabinet. It supports the same functionality as the SIU. A TCU supports 8 E1/T1 that can be used for Abis lower or Abis upper (ML-PPP). The TCU is supported from G12B.

- The hardware called Baseband T605 is an STN made to fit into the RBS6000 cabinet or to be used as a standalone STN. Baseband T605 enables efficient co-transport of GSM, WCDMA, LTE, and CDMA traffic over IP on shared backhaul for efficient use of the transport network. A Baseband T605 supports 8 E1/T1 that can be used for Abis lower. The Baseband T605 is supported from GSM RAN G14B.

- A Baseband Radio Node is a multi standard capable node including BTS, NodeB and/or eNodeB functions. The node terminates the IP protocol that is used on Abis upper. The Baseband Radio Node is supported from GSM RAN G16B.

## 2.3 Hardware Capabilities

*Table 1 Capabilities for required and optional hardware*

| Hardware | TGs | TRXs | Erlang | Interface type support |
|---|---|---|---|---|
| 1xSIU (STN) | 6 | 36 - 72(1) | 700 - 900 | IP over E1/T1, Ethernet (1Gbit/100Mbit) |
| 1xTCU (STN) | 6(2) | 36 - 72 (1) | 700 - 900 | IP over E1/T1, Ethernet (1Gbit/100Mbit) (2) |
| Baseband T605 (STN) | 6 | 36 - 72 (1) | 700 - 900 | Ethernet (10Gbit/1Gbit) |
| 1xRBS2409 (Pico) (STN/BTS) | 1 | 2 | 30 | Ethernet (100Mbit/10Mbit) |
| Baseband Radio Node (5212) | 48 | 24 | | Ethernet (10Gbit/1Gbit) |
| Baseband Radio Node (5216) | 48 | 48 | | Ethernet (10Gbit/1Gbit) |

*(1) Regardless of traffic model, 36 TRXs are supported. Up to 72 TRXs can be handled.*

*(2) Using IP over E1/T1 may limit the number of TGs that can be handled due to limited number of E1/T1.*

# 3 Technical Description — Abis over IP Transport

## 3.1 General

The transport network described in this chapter refers to the BSS nodes and their signaling, speech (CS), and data (PS) traffic. The characteristics of the transport network is described in Section 4 on page 25. The O&M network, associated with configuration, supervision and software management, is described in Section 5 on page 27.

With the feature Packet Abis over IP, all traffic types are sent as IP packets on a Wide Area Network (WAN) instead of as bit streams on dedicated time slots as in TDM (also referred to as the “classic Abis”). The IP network illustrated by a cloud in Figure 1, is often bought from an Internet Service Provider (ISP) as a set of public IP services, regulated by a Service Level Agreement (SLA).

The feature Packet Abis over IP uses the same packetized Abis framework as Packet Abis over TDM (see Reference [2]). Packetizing refers to the process of converting bit streams to IP packets. “Packet Abis” is the term used for a packetizing Abis interface, regardless of whether it is implemented with the feature Packet Abis over TDM or the feature Packet Abis over IP.

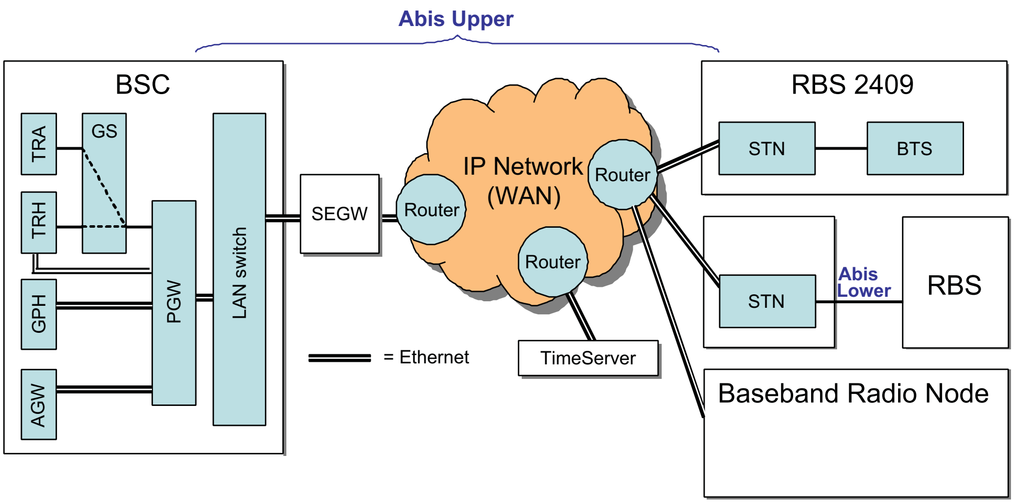


*Figure 1 Transport Network of Abis over IP*

In the TDM based Abis, the BTSs are synchronized to the transport network using information in the E1 or T1 frame structure. The Ethernet transport network in Abis over IP is instead asynchronous and cannot provide synchronization to BTS, which is why BTS has to be synchronized in another way. The standard solution is to use a TimeServer for synchronization of Abis over IP. More alternatives and details are found in Reference [7] and Reference [32].

## 3.2 Basic Concepts

### 3.2.1 Abis Path

The term Abis Path changes meaning with Packet Abis. Previously, it mirrored the physical 64 kbps path between BSC and BTS, but in Packet Abis it becomes a virtual path end-to-end where some parts are still 64 kbps but others have higher rate.

### 3.2.2 Super Channels

The Super Channel (SC) concept applies to Abis Lower, the interface between STN and BTS. Note that for a Baseband Radio Node there is no Abis Lower and therefore no SCs.

In BSC, SCs still exist as objects for O&M and load regulation purposes. One SC is one E1/T1 link where 64 kbit/s consecutive Abis time slots are used as a wideband connection for sending LAPD frames. For E1s, 30 or 31 TSs are used, while 24 TSs are used for T1s. SCs are a set of contiguous time slots on a TDM link between one port in STN and one BTS port. The relationship between TRXs and SCs is fixed. One TRX cannot transmit over different SCs. One TG can communicate via up to four SCs, and the SCs connected to one TG are grouped into a Super Channel Group (SCGR).

### 3.2.3 Protocol Layers

Protocol layers are protocols nested in a protocol stack, as shown in Figure 2. The principle is that data content of a lower layer is filled with the higher layer header and its data content. The first layer (L1) is just a specification of how digital data are sent. From layer 2 (L2) and upwards, each layer consists of both a header and a data content part.

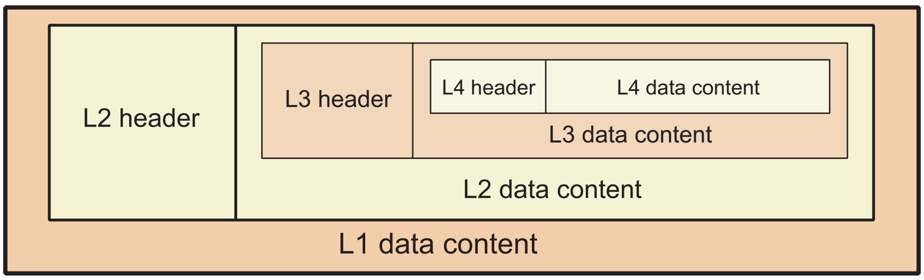


*Figure 2 Nested Protocol Layers*

Some of the protocols used in Abis over IP are following an international standard, while others are Ericsson proprietary protocols. The Open System

Interconnection (OSI) architecture, containing seven protocol layers, is not applicable on Abis over IP, but used as a comparison. The following are the lowest three layers of the OSI model:

**Layer 1** The *Physical Layer* provides mechanical, electrical, functional and procedural resources for transmission of bits. It contains functions for converting data into signals compatible with the transmission medium.

**Layer 2** The *Data Link Layer* concerns inter node connections. The data link layer provides an error free point-to-point circuit between network nodes. The layer contains resources for error detection and correction, flow control and re-transmission. It ensures that the messages are sent in the correct order, without errors or duplication. Layer 2 headers are exchanged in each node they pass through the network.

**Layer 3** The *Network Layer* isolates the upper layers from routing and switching functions in the network. The functions within the network layer establish, maintain, and release connections between nodes in the network, and handle addressing and routing of messages. Layer 3 headers contain the end destination and are not exchanged under their way to the destination.

The two architectures described in this document are IP over Ethernet and IP over PPPoE. Also E1/T1 based transport, using ML-PPP as the layer 2 protocol, is supported in the BSS nodes, but described in Reference [3] .

### 3.2.4 IP over Ethernet Protocols

The protocol layers used in Abis over IP for Ethernet based transmission are shown in Figure 3. Layer 1 is visualized as connection lines between the nodes. Nevertheless, E1 and T1 has some physical characteristics as well.

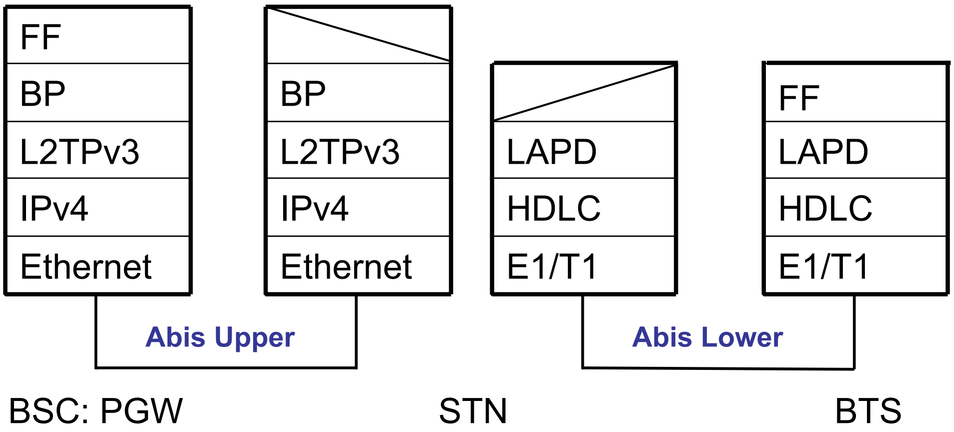


*Figure 3 Protocol View, STN case*

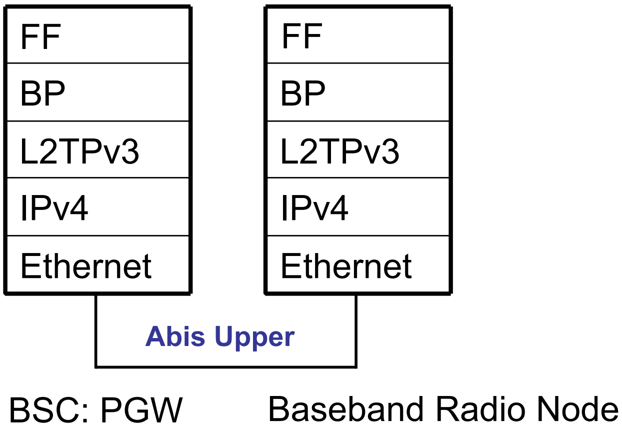


*Figure 4 Protocol View, Baseband Radio Node case*

On Abis Lower either the E1 or the T1 protocol is used. An E1/T1 channel is used as an SC between STN and BTS. Up to 4 SCs are connected per TG.

HDLC is a protocol that operates at layer 2. All frames start and end with the flag ='01111110'. The frames also contain a check sum. HDLC contains many other protocols, like LAPD, LAPB and SDLC.

LAPD is a layer 2 protocol used for signaling on the Abis Lower interface, but also internally in the BSS nodes. LAPD has been used for differing purposes, one of them is Abis signaling. LAPD provides a transparent connection between two layer 2 peers. The frame contains address, data information, among other things. For further details see ITU-T recommendation Q.921 (I.441).

IP is a layer 3 protocol, containing the end destination in its header.

L2TP is a tunneling protocol that hides information from the IP network. L2TP uses transport sessions to tunnel LAPD frames and uses a control connection to manage the connection.

The Bundling Protocol (BP) is an Ericsson proprietary standard and packs a number of LAPD frames into one IP packet. Each LAPD frame contains one FF frame.

On FF level the Abis communication format equals classic TDM transmission, with RSL frames, OML frames, speech frames, and data frames (GPRS).

## 3.3 Transport Flow

### 3.3.1 Downlink Transport

In the downlink direction, Abis over IP starts in the Packet Gateway (PGW) in BSC. Signaling, CS (primarily speech), and PS (data) are handled in somewhat different ways in the PGW, but all in the same RP. The different traffic types arrive to the PGW in different formats. The TRH sends signaling in the form of LAPD frames via the Group Switch (GS) interface or Ethernet interface for EVO 8200 when a GS is not present. The TRA formats speech (or CS data) to TRAU frames and sends them streamed via the GS interface. Speech frames can also be received in a frame format directly from the AGW if Ethernet based AGW - PGW transmission is used. The GPH (the RP parts of the PCU) transforms the PS data to Ethernet packets and sends them over Ethernet. The PGW compresses and converts all the traffic types to IP packets on a bundled format. The bundling algorithm is described separately in Section 3.3.3 on page 12. Internally, the IP packets are sent over Ethernet to the BSC IP network interfaces. The BSC IP network interfaces read the IP headers and decides where to send the packets. They are sent via Ethernet out on the IP network of Abis Upper.

The targeted STN or Baseband Radio Node receives the IP packets on its WAN interface and unpacks them to LAPD frames. In the STN case, STN sends the LAPD frames over Abis Lower to the right BTS via a configured super channel. The TEI value in the BP header specifies the targeted TRX, and each TRX is mapped on a super channel. The super channels are described in Section 3.2.2 on page 8.

### 3.3.2 Uplink Transport

In the uplink direction, the reverse process to the above evolves.

In the STN case, STN receives LAPD frames from a BTS via the configured super channel on Abis Lower. The super channels are described in Section 3.2.2 on page 8.

STN or Baseband Radio Node bundles frames belonging to the same TG. The bundling algorithm is described separately in Section 3.3.3 on page 12. The IP packets are sent through the WAN interface out on the IP network of Abis Upper.

In the other end, the BSC LAN Switch/BSC NWI-E receives the IP packets on its Ethernet interface and routes them to the PGW. The PGW unpacks the IP packets and uncompresses the data contents, as well as converts them to the right format. Signaling is formatted to LAPD frames and sent to the TRH either via the GS or directly over Ethernet if no GS is present. Speech (or CS data) is converted to TRAU frames and streamed via the GS to the TRA or sent as frames directly to the AGW if Ethernet based PGW - AGW transmission is used. PS data is sent as Ethernet packets directly to the GPH.

### 3.3.3 Bundling

Bundling of IP packets is not included in the L2TP standard for HDLC tunneling, why Ericsson has developed a proprietary addition to the standard. The idea is to decrease the IP overhead by grouping several frames belonging to the same TG in a common IP packet. The operator can configure the bundling algorithm to prioritize the most important network characteristics: bandwidth, delay, or packet loss rate. This configurability per PGW-STN/Baseband Radio Node link makes it easier to achieve the most suitable transmission on a per site basis.

The bundling algorithm collects frames, strips their headers, and gives them a common IP header. An IP packet is sent either when it is filled with complete frames up to the maximum packet size, or when the bundling time (maximum collection time) has elapsed. The bundling time is measured from putting the first frame into the IP packet.

The larger IP packet size, the more time it takes to fill one packet. Large IP packets also increase the probability that a packet is discarded due to transmission related bit errors. Small IP packets decrease the delay, in proportion to the time needed to collect frames, but increase the IP/L2TP overhead. Similarly, a long bundling time decreases the IP/L2TP overhead, depending on the amount of traffic, but increases the delay, while a short bundling time increases the overhead and decreases the delay. The parameters of maximum packet size and bundling time are configured separately for the uplink and downlink directions.

In the STN case, the bundling algorithm is configured in a bundling profile called LBG. By default, one profile (LBG 0) contains all traffic types, but if the operator would like to assign different configurations for different groups of traffic types, an LBG profile must be created for each bundling group. The LBG profile specifies what traffic types to be bundled into the same IP packet and their priority value, called DiffServ. Diffserv, and its use to assign different quality of service to different traffic types, is described in Section 3.5 on page 14. There is always one bundling group per LBG.

In the Baseband Radio Node case, the bundling algorithm is configured per traffic type (SAPI). Traffic types that have the same bundling configuration forms a bundling profile. By default, there is one profile for signaling, one for CS and one for PS. If the operator would like to assign different configurations for different groups of traffic types, this configuration can be changed. The profile specifies what traffic types to be bundled into the same IP packet and their priority value, called DiffServ. Diffserv, and its use to assign different quality of service to different traffic types, is described in Section 3.5 on page 14. The bundling groups are automatically created and updated and the number of bundling groups varies depending on configuration and the Baseband Radio Node's TRX allocation in hardware.

## 3.4 Abis over IP Initiation

### 3.4.1 BSS Start-Up

In the STN case, BTS starts scanning for the CF-link on Abis Lower as soon as it is switched on.

The operator configures BSC to Abis over IP from OSS. BSC initiates the IP stack to start listening for L2TP control connection establishment requests from the STN or Baseband Radio Node. The operator configures L2TP control connection IP addresses both to the STN/Baseband Radio Node and the PGW. The IP link is initiated, which means that the control connection, as well as the transport sessions on the L2TP protocol, are started.

### 3.4.2 STN/Baseband Radio Node Start-Up

STN/Baseband Radio Node starts establishing a control connection to the PGW using its pre-configured IP address to PGW. If the response is a redirect, a new control connection IP address is sent back, and STN/Baseband Radio Node tries to establish a control connection to the new IP address. From now on, STN/Baseband Radio Node uses the dedicated IP address to the PGW configured to handle this TG. In the case no response is received after a configurable number of retries, STN/Baseband Radio Node raises an alarm to OSS.

In the STN case:
When the control connection is established, STN receives a transport configuration message from the PGW which tells the TG Transport what transport sessions to establish. For each transport session, STN sends a request. The PGW replies with an acknowledgement. STN repeats the request for a transport session until an acknowledgement is received from the PGW or its state changes to not being able to execute traffic. If the acknowledgement is not received after a configurable number of retries, STN raises an alarm to OSS.

In the Baseband Radio Node case:
When the control connection is established, a bundling group dedicated for Sector CF (SCF) OML signaling is established. This SCF OML link is used to send SCF messages, including bundling information. Based on the information

received from the BSC and the internal configuration in the Baseband Radio Node, the Baseband Radio Node decides what transport sessions to establish. This is returned to the BSC. For each transport session, Baseband Radio Node sends a request. The PGW replies with an acknowledgement. Baseband Radio Node repeats the request for a transport session until an acknowledgement is received from the PGW or its state changes to not being able to execute traffic. If the acknowledgement is not received after a configurable number of retries, Baseband Radio Node raises an alarm to OSS.

The above mechanisms imply that the IP addresses used by the PGW are reachable from STN/Baseband Radio Node. If the IP address plan somewhere in the network between BSC and STN/Baseband Radio Node prohibits this, the operator is forced to use some VPN mechanism, for example IPSec, to hide the address plan of the transport network from the Abis application.

### 3.4.3 BSC Start-Up

In BSC, each PGW-RP or CTH-RP is configured with its own IP address, but there is also always one common IP address active on one of the RPs. This initial IP address enables an STN/Baseband Radio Node to establish its control connection even if the RP assigned to handle the TG is out of service (provided that the recommended redundancy has been configured). This dynamic connectivity is part of the PGW redundancy described in Reference [4] and Reference [5].

When the PGW receives a control connection establishment request at the initial IP address, the received host name is mapped to the IP address of the RP that is currently assigned to handle the TG. PGW replies with a control connection redirect message, informing the STN/Baseband Radio Node about which IP address to contact. When the right PGW receives a control connection establishment request, the source IP address of the message is stored for this TG, and an acknowledgement is returned. If an alarm had been raised earlier, it is ceased when the STN/Baseband Radio Node receives the acknowledgement, and the periodic supervision procedure is started. In the STN case, a transport configuration message is sent to the STN, indicating what transport sessions to establish. In the Baseband Radio Node case, the BSC waits for the SCF OML link to be established, as described in Section 3.4.2 on page 13.

## 3.5 Quality of Service

Quality of Service (QoS) is used to identify traffic with different requirements on quality of service, and is based on the IP standard. It is implemented with a priority tag, DiffServ, in the IP header of each bundled IP packet. In the normal case, there are no strong benefits to give different traffic types different priority with DiffServ, but when the bandwidth is limited and there is a risk for overload, typically in satellite transmission, separate priorities are recommended. However, not all IP services support the use of DiffServ.

The traffic type of a LAPD frame is defined by its SAPI value. In BSS the following SAPI values are used over Abis: 0 for RSL, 10 for Speech, 11 for CS Data, 12 for (E)GPRS, and 62 for OML. In addition to the traffic types supported by the BSC, the STN/Baseband Radio Node also supports L2TP control signaling, synchronization (NTP/PTP) towards the TimeServer and O&M signaling towards OSS. SAPI values and the use of the check sum CRC32 are identically configured for the uplink and downlink directions. The configuration of QoS is described in Section 6.6 on page 56.

## 3.6 Load Regulation on Abis Upper

### 3.6.1 General

The concept “load regulation” is here used for both overload prevention and overload handling. “Overload prevention” refers to load regulating actions initiated at high transmission load. Their purpose is to decrease the risk of overload. “Overload handling” refers to the load regulating actions initiated in an actual overload situation. Their purpose is to reduce traffic and decrease the risk of losing whole cells or TGs.

The overload handling is an algorithm that detects "probable" overload, i.e. there can be situations where overload is triggered when there is no actual overload. One examples of this can be short bursts of high drop rate. Due to this it is recommended to turn overload handling off if Abis is deployed on a transmission with a bandwidth that is high enough for any expected traffic.

BSC has the main responsibility to evaluate and order actions to counteract congestion, and regulates both uplink and downlink transmission simultaneously.

During normal operation, with a correctly dimensioned IP network and with the solutions for IP traffic level control already implemented in the system, there is no risk of losing a TG due to increased delay and lost control signaling. However, if the IP network is under-dimensioned or changes to the IP network capacity occur (for example due to damage or reparation), the result may still be an overload situation.

When the bit rate on a link somewhere in the IP cloud between BSC and the STN or Baseband Radio Node is higher than the available transmission capacity, the router or switch queue builds up and increases the transmission delay rapidly. The control signaling (RSL and OML) is lost if the delay grows too long or too many consecutive retransmissions occur.

Situations where overload problems can occur:

- **Traffic peak:** In this case, the transmission capacity is constant. The bit rate of the Abis over IP flow is less than the transmission capacity. Suddenly there is a short peak of a bit rate larger than the available capacity.

- **Short burst of lower transmission capacity:** In this case, the bit rate of the Abis over IP flow is constant. The capacity of the transmission is lowered for a short time.

- **Engineered bandwidth larger than actual bandwidth:** In this case, the engineered bandwidth (explained in Section 3.6.2 on page 16) is set too high. If the bit rate of the Abis over IP flow is larger than the transmission capacity during busy hours, the TG may lose traffic capacity periodically.

### 3.6.2 Load Regulation Triggers

Utilized bandwidth, packet loss and retransmission are indicators of load problems. The operator configures threshold values for most of these indicators to trigger load regulation actions at suitable points in time. Recommended configurations are found in Section 7.1.5 on page 71.

Utilized bandwidth is not the actually used bandwidth since it is measured in percentage of the engineered bandwidth (**MBWDL** and **MBWUL**). If there are disturbances in the transmission network that temporarily reduce the actual bandwidth, the measured utilized bandwidth might not trigger any load regulation actions. The engineered bandwidth is instead aimed at situations when the bandwidth approaches the SLA bandwidth.

In the single TG case the engineered bandwidth should be set close to the SLA bandwidth but not exceeding it.

However, the operator is not recommended to set engineered bandwidth close to SLA bandwidth in the multi TG case, since engineered bandwidth is configured per TG. If all TGs are of equal size, the engineered bandwidth intuitively would be set to SLA bandwidth divided by number of TGs, but such a configuration would make load regulation actions trigger for a TG even if there is still common bandwidth available. A higher engineered bandwidth admits a TG to take advantage of situations where not all TGs approach overload at the same time. For a normal case the total engineered bandwidth can be over dimensioned with about 5% per added TG, i.e. 105% of the SLA bandwidth for the two TG case and up to 125% for the 6 TG case. All overload thresholds that relate to the engineered bandwidth should in this case be decreased so that they still reflect the same bandwidth as if no over dimensioning were applied. This is done to avoid the risk of getting overload prevention triggers at a level higher than the actual available bandwidth. If there is a good knowledge of the traffic variations over time in different TGs a more or less aggressive dimensioning and threshold setting can be used.

The downlink and uplink engineered bandwidth, are used indirectly to trigger overload prevention, since the threshold values are set to percentage of engineered bandwidth. Also, **MBWDL** and **MBWUL** directly trigger overload handling, if the utilized bandwidth exceeds the engineered bandwidth in any direction.

The occurrence of packet loss in Abis over IP may be an indication of overload. Packet loss is defined as a missing sequence number in the packet transmission. Even a packet arriving out of order is regarded as lost. The triggering parameter is **OVLTH**. In addition to this the PGW also has a fixed minimum traffic level below which this overload trigger is not activated. This minimum level is 4 speech calls and 150 kbps UL + DL transmission on Abis. Packet loss is detected in both uplink and downlink direction. PGW counts uplink packet loss. STN or Baseband Radio Node counts downlink packet loss, and reports the information to PGW. In order to get this function to work properly, and not give false overload indications, it is important to configure the threshold to a value above the drop rate of the Abis transmission during normal operation (not overload). Note that it is the peak drop rate that should be considered.

Retransmission of RSL signaling frames is also used as an indication of overload, but it is detected only in the downlink direction. When the same connection has tried to send a signaling frame four times without success, overload handling is triggered.

### 3.6.3 Overload Prevention

Overload prevention consists of the two features Abis Triggered HR Allocation and Fullrate AMR on 8 kbps Abis, both triggered by thresholds for utilized bandwidth (in percentage of engineered bandwidth). The triggered actions are summarized together with their trigger points and parameters in Table 2.

The feature Abis Triggered HR Allocation initiates two separate overload prevention actions for half rate capable terminals. One action is to force new speech calls to allocate half rate channels. The other is to force ongoing full rate speech calls to move to half rate through hand-over. The two actions are triggered by independent thresholds (Table 2). By the help of quality triggers, high quality calls are the first ones to be forced to half rate. If the feature Speech Quality Priority is used, subscribers with lower priority levels get half rate before subscribers with higher priority levels. When the bandwidth utilization returns below another threshold value (Table 2), half rate calls start moving back to full rate again. Configuration issues are treated in Section 7.1.5 on page 71.

The feature Fullrate AMR on 8 kbps Abis limits the set of codecs used when Adaptive Multi Rate (AMR) is active. Originally, AMR is a technique to select a codec on the air interface depending on current signal quality. The worse quality on the air interface, the more redundancy should be used on the air interface. This is achieved by using codecs with a lower rate and add protection coding on the air interface. In this way the air interface transmission is maintained on full rate but on Abis interface only the lower rate equal to a half rate is used. In the feature Abis over IP, this codec rate reduction is used to prevent overload on Abis. When the threshold value is exceeded (Table 2), connections are forced to use more efficient codecs. The maximum codec rate is limited to 7.4 kbps on Abis for AMR full rate calls on the air interface. Configuration issues are treated in Section 7.1.5 on page 71.

*Table 2 Overload Prevention Trigger Points and Actions*

| Trigger Point | BSC Parameter | Actions |
|---|---|---|
| The bandwidth utilization exceeds the threshold. | **SDHRAABISTHR** (configurable threshold) | Feature Abis triggered HR allocation: New CS Connections are forced to allocate half rate (not including AMR-WB) |
| The bandwidth utilization exceeds the threshold. | **SDFRMAABISTHR** (configurable threshold) | Feature Abis triggered HR allocation: Existing full rate CS connections start moving to half rate (including AMR) |
| The bandwidth utilization falls below the threshold. | **SDHRMAABISTHR** (configurable threshold) | Feature Abis triggered HR allocation: Half rate CS connections start moving (back) to full rate (including AMR) |
| The bandwidth utilization exceeds the threshold. | **SDAMRREDABISTHR** (configurable threshold) | Feature Fullrate AMR on 8 kbps Abis: AMR codec is restricted to max AMR 7.4 kbps over Abis but uses AMR FR on the air interface |
| The bandwidth utilization exceeds the threshold. | **SDHRAABISTHRWB** (configurable threshold) | Feature Abis triggered HR allocation: New CS Connections that are AMR-WB capable are forced to allocate half rate |

### 3.6.4 Overload Handling

Even if the above overload prevention actions are triggered, there is still a risk of overload. The overload handling actions are presented in Table 3. Also overload prevention actions are included in overload handling as long as those features are enabled, which means that they are triggered even if their own thresholds have not been exceeded.

#### 3.6.4.1 Overload triggers

There is three different overload triggers, IP packet drop rate, RSL signaling drop and max available bandwidth.

When the IP packet loss threshold **OVLTH** in a bundling group is exceeded, the PGW initiates overload handling (see details in Table 3). The overload handling is triggered as soon as the threshold is exceeded. In the downlink direction, the trigger is based on notifications from STN or Baseband Radio Node. The PGW compares the reported packet loss to the **OVLTH** of the valid bundling profile and triggers overload handling if that **OVLTH** has been exceeded. **OVLTH** both triggers and stops overload handling, why it is combined with timers to avoid the “ping-pong effect”. Configuration issues are treated in Section 7.1.5 on page 71.

RSL signaling drop detection is active if **IPOV** is set to on and **LDEL** is set to 1. Overload handling actions are initiated when an RSL signaling frame in a connection has been sent four times (3 timeouts, see details in Table 3). Then, PS and CS connections are rejected. More configuration details are found in Section 7.1.5 on page 71.

The third trigger mechanism is exceeding engineered bandwidth in uplink or downlink, set by parameters **MBWUL** and **MBWDL**, see Section 7.1.3.2 on page 67. Overload handling actions are initiated when the load on the TG exceeds any of these parameters. Then PS data scheduling will be omitted gradually to decrease the load downlink.

#### 3.6.4.2 Overload actions

**Omit PS scheduling**

This is the lowest level of overload action regarding impact on end user performance. This action omits a ratio of PS data scheduling to decrease the load. This ratio will gradually increase until all PS data scheduling is omitted if the overload persists. If the overload situation is ceased the ratio of omitted PS data scheduling will gradually decrease until no PS data scheduling is omitted.

Note that omission of PS scheduling also can be triggered by high CPU load in the PGW/CTH RP (see Reference [4] and Reference [5]).

If the overload situation persists and all PS data scheduling is omitted and the next level of actions are triggered, this is rejection of new PS connections and new CS connections.

**Reject of PS connection**
No new PS connections are allowed and the existing PS connection will be gradually terminated depending on how long this action is continued. In this action all of the PS scheduling is omitted.

**Reject of CS connection**
Number of CS connections are limited to a level close below the number of connections established when entering this action. New CS connections can be allowed if an existing is ended and the current number of connections are below the limit. If the overload persists the limit is continuously decreased. If the overload is ceased the limit is gradually increased until the limit is equal or above the starting number of connections. At this point new PS and CS connection are allowed and the Omit PS scheduling action is started.

Table 3 defines how triggers and actions interact.

*Table 3 Overload Handling Trigger Points and Actions*

| Trigger Point | BSC Parameter | Actions |
|---|---|---|
| The packet loss ratio in a bundling group exceeds the threshold. | **OVLTH** in a bundling profile containing PS (but not CS) traffic (configurable threshold) | Overload prevention(1) + Omit PS-scheduling(2) |
| The packet loss ratio in a bundling group exceeds the threshold. (non QoS) | **OVLTH** in a bundling profile containing CS (and possibly PS) traffic (configurable threshold) | Overload prevention(1) + Omit PS-scheduling(2) |

*Table 3 Overload Handling Trigger Points and Actions*

| Trigger Point | BSC Parameter | Actions |
|---|---|---|
| The packet loss ratio in a bundling group exceeds the threshold. (QoS) | **OVLTH** in a bundling profile containing CS traffic (configurable threshold) | Overload prevention[(1)] + RejectPS[(3)] + RejectCS[(4)] |
| The bandwidth utilization exceeds the engineered bandwidth. | **MBWUL**, **MBWDL** (configurable values) | Overload prevention[(1)] + Omit PS-scheduling[(2)] |
| 3 consecutive RSL signaling frames are retransmitted. | **IPOV** (on/off), **LDEL** (0/1) | Overload prevention[(1)] + RejectPS[(3)] + RejectCS[(4)] |
| 4 consecutive RSL signaling frames are retransmitted. | **IPOV** (on/off), **LDEL** (0/1) | Overload prevention[(1)] + RejectPS[(3)] + RejectCS[(4)] |

*(1) The two overload prevention features described in Section 3.6.3 on page 17 are initiated even if their thresholds have not been exceeded.*
*(2) Omit PS-scheduling = A number of PS-scheduling are omitted to decrease the load. If this action isn't enough and all PS-scheduling are omitted the next action will trigger RejectPS[(3)] and RejectCS[(4)]*
*(3) RejectPS = New PS connections are rejected on all PDCH types and all PS-scheduling are omitted.*
*(4) RejectCS = New CS connections are rejected.*

## 3.7 Load Regulation on Abis Lower

In G14B load regulation and traffic shaping functions were introduced on Abis Lower for Packet Abis over IP. The same functionality has previously been used by Packet Abis over TDM. For a detailed description of this functionality see the user description for Packet Abis over TDM Reference [2].

In the STN there is also a shaping function with strict priority between signalling, CS and PS that discards frames that cannot be fitted into the super channels.

## 3.8 Transport Security

When the IP services are bought externally, the security aspects become critical. A number of measures can be taken to reduce the increased connectivity in an IP network solution:

- Implement access control lists in site routers (RBS, BSC and OSS sites).
- Implement firewalls in front of sensitive nodes, such as BSC and OSS, or at interconnection points between the Abis and core IP networks.
- Limit connectivity in the IP network by implementing some type of network virtualization.

- Implement IPSec tunnels from the RBS site to BSC and OSS sites or to a secure IP backbone.

The implementation of the above measures is made in site and network equipment not part of the Abis over IP solution. If the IP access network is not trusted, IPSec can be used to create secure tunnels from the RBS site to the core IP network or the BSC and OSS sites.

As a consequence of the need described above, SEGWs will be deployed in BSC and RBS sites to implement IPSec tunnels between BSC and RBS sites. The following terms are used:

- **RAN SEGW** is a security gateway (firewall) located at the BSC site to provide a secure IPSec tunnel towards the RBS site through an external, unsecure transport network. This functionality is provided by an external equipment.

- **RBS SEGW** is a security gateway (firewall) located at the RBS site to provide a secure IPSec tunnel towards the BSC site through an external, unsecure transport network. This functionality is provided by the STN and Baseband Radio Nodee.

The capacity and scalability requirements in RBS SEGWs are not as demanding as in RAN SEGWs.

SEGWs are network nodes optimized for security handling. They can manage tens, hundreds or thousands of IPSec VPN tunnels. Additionally, they offer extensive firewall capabilities.

The use of an IPSec VPN for Abis over IP traffic while in transit through public or semi-public transport networks is assumed as part of the Abis over IP Ericsson implementation.

The IPSec protocols suite provides different levels of securing IP communication by providing authentication, integrity and/or confidentiality. It is recommended to analyze for each specific case the most appropriate set of parameters according to a full threat analysis depending on the operator’s policies and the type of transport network.

Independent networks: several VPNs defined in the transport network that enables traffic separation from the RBS site into several different IPSec tunnels carrying different types of traffic (NTP, Abis user/control plane, and O&M).

In the case of having a common network scenario (a VPN defined in the transport network that carries all traffic from the RBS site to the BSC site) no traffic separation is used through the transport network as just one IPSec tunnel is assumed between the RBS and the BSC site. In that way, the BSC site security domains cannot be extended through the transport network to the RBS site, as the IPSec tunnel constitutes a single security domain. It is, therefore, recommended to use the firewall capabilities on those RAN SEGWs, to control traffic between this IPSec security domain and the site Security Domains (used for Abis traffic, O&M traffic and NTP traffic). For an independent networks scenario several VPNs enable traffic separation from the RBS site into several different IPSec tunnels carrying different types of traffic (NTP, Abis user/control plane, O&M), 3 IPSec tunnels physically terminated on different equipment (SEGWs) has to be considered over the transport network. The three IPSec tunnels constitute 3 different Security Domains. The SEGWs on each independent network controls the traffic between each IPSec service domain to the correspondent Site service domain.

On the RBS side it is not possible to provide traffic separation between Abis traffic, STN/Baseband Radio Node management traffic and NTP traffic. Therefore it constitutes a single service domain, the RBS service domain. Also an RBS SEGW is deployed for the IPSec termination on the RBS site. For the independent network scenario, firewall capabilities on the RBS SEGW should also be used to control traffic from the RBS service domain to the three different IPSec service domains. When considering the common network scenario, only one IPSec tunnel is considered through the transport network that constitutes a separate service domain, the required control from the RBS service domain to the IPSec service domain is provided by the RBS SEGW.

All O&M of the STN and Baseband Radio Node uses SSH (TLS is also supported by Baseband T605 and Baseband Radio Node) and SFTP. User authentication is mandatory for all O&M connections.

More information on the use of SEGWs and IPSec is found in Section 6.8 on page 59. Managing of IP security for Baseband Radio Node is described in Reference [31]. General IT security issues are described in User Description, GSM RAN IP Security, Reference [9].

## 3.9 IP over PPPoE

With the PPP over Ethernet (PPPoE) solution the security aspect is taken into consideration by using IPSec tunneling through the public part of the network (Figure 5). The RBS 2409 supports PPPoE in a DSL access network (for example ADSL). PPPoE provides the ability to connect a network of hosts to a remote access point. PPPoE is used between STN and the Broadband Remote Access Server (BRAS), where the BRAS acts as a server and the RBS 2409 acts as a client.

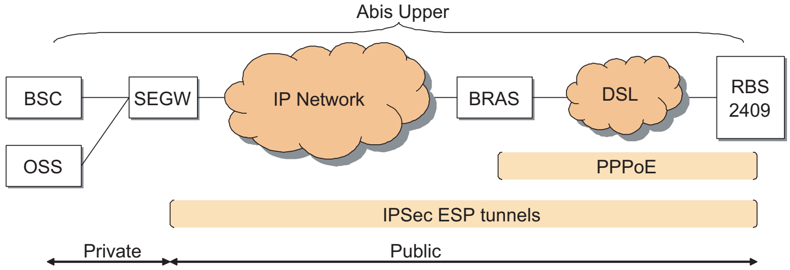


*Figure 5 PPPoE Network*

Authentication is part of the Point to Point Protocol (PPP). This means that the peers can authenticate each other before allowing Network Layer protocols to transmit over the link.

# 4 Technical Description — Transport Network Characteristics

The characteristics of the transport network and its impact on the end user performance is described in Reference [29]

# 5 Technical Description — Operation and Maintenance

## 5.1 General

This chapter describes the O&M Network and transmission related to configuration, supervision and software management. The transport network and its characteristics are described in (Section 3 on page 7 and Section 4 on page 25).

## 5.2 O&M of BSC

O&M, such as software management and supervision of the PGW, is handled in the same way as other RP based applications (for example GPH and TRH) by BSC functions. For detailed information on parameters see Section 7.1 on page 63.

## 5.3 O&M of SIU and TCU

### 5.3.1 General

O&M support for the STN is implemented in OSS and the STN is seen as a separate network element. The OSS support depends on the OSS revision used.

The O&M transmission links are shown in Figure 6. Abis over IP does not affect how O&M is managed for BSC, BTS and the BSC LAN Switch/BSC NWI-E. The TimeServer, SEGW and STN are managed by OSS as stand alone network elements. They are configured directly from OSS. Both the BSC LAN Switch and BSC NWI-E are also configured directly from OSS, while BTS is configured via BSC from BSM in OSS. Software management of STN is managed via the SMO application in OSS. Besides the OSS applications for O&M, the operator may use MML or CLI sessions to send O&M commands over SSH or Telnet.

The operator can monitor the amount of IP traffic to each base station, as well as the quality of the network to each BTS.

For further information on O&M see Reference [13] and O&M documentation in relevant node CPI.

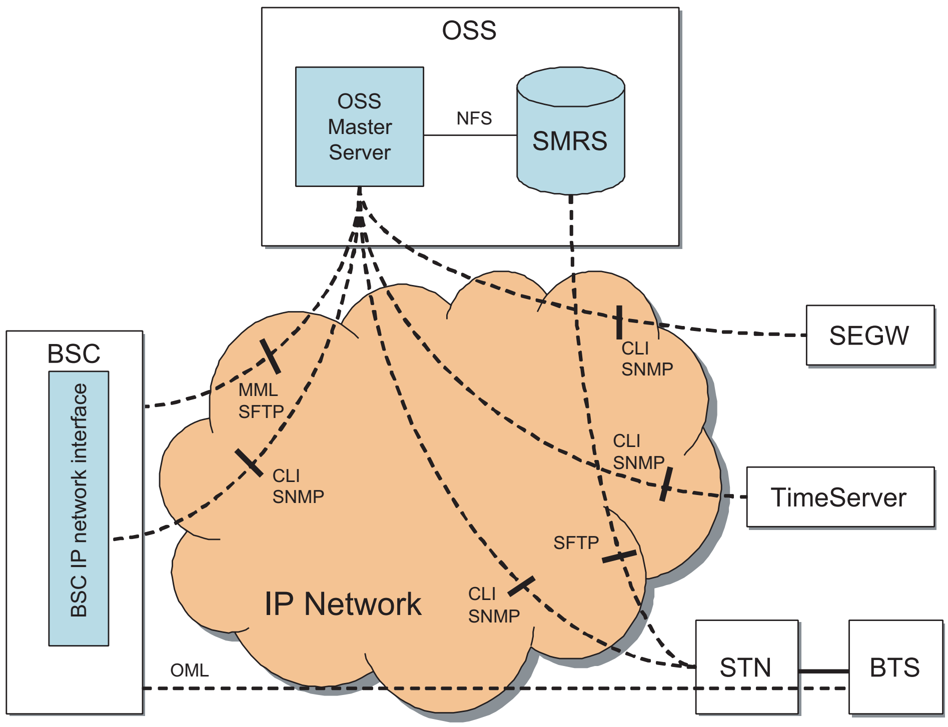


*Figure 6 OSS O&M Interfaces*

## 5.3.2 Initiation of the STN-OSS link

The STN identifies itself to the O&M system at installation. Before installation, STN is configured with its own IP parameters and host name, as well as an IP address to the O&M system. STN sends its IP address and host name to OSS.

STN sends a wake-up message (SNMP trap) to OSS at start-up. The wake-up message contains the STN IP address and host name. When OSS receives the message, it recognizes the host name and connects the OSS network element to the corresponding IP address. At this point the periodic supervision of the STN starts (heartbeat). Now, STN can be configured from OSS, if needed.

## 5.3.3 Supervision

Supervision is in this document the concept used for Fault Management and Performance Management. A mechanism for supervision of Abis Upper exists, and is handled by STN. STN sends a keep-alive message to BSC on a periodic basis (the period is configurable). If no answer is received, STN raises an alarm to OSS. STN tries to reestablish the connection, and when it succeeds, the alarm ceases.

Supervision of Abis Lower is described in Reference [15] (valid also for co-located BTSs).

### 5.3.4 Configuration Management

The STN persistently maintains the current configuration. The configuration is performed using set and get operations via CLI or Bulk CM operations using SFTP towards the SMRS server.

The Bulk CM is formatted into an XML file that contains “sub-operations”, for example create, delete and modify of MOs and attributes. This configuration file is stored in a configuration file repository on the SMRS servers. The bulk operations are available via SSH and CLI. Administrative information about the file repository is provided in the operations.

For detailed information on parameters see Section 7.2 on page 77.

### 5.3.5 STN Parameter Configuration from BSC

BSC transfers the following Abis over IP related parameters to STN after connection establishment:

- DiffServ code per bundling group
- List of SAPI values defining a bundling group
- Bundling parameters: bundling packet size and bundling time
- Checksum ON or OFF
- BSC IP address for the transport session, per bundling group
- TRX to SC mapping (via TEI)

### 5.3.6 Fault Management

Alarms and stateless events are generated from STN and transferred to OSS using SNMP V2C, based on STN defined MIB. Alarms and events are formatted based on 3GPP Fault Management, Alarm IRP, Information Service, see O&M documentation in relevant node CPI. A Heartbeat event is generated by STN to OSS using a configured time interval.

### 5.3.7 Performance Management

STN records different performance events in separate counters. The sampling period is 15 minutes. A file is generated with the performance counters and their values. The formatted file is transferred to a Performance File Repository using SFTP.

Administrative information about the location of Performance File Repository (URL, username and password), report period, and so on, are provided as a part of the STN configuration. The administrative information is reachable via

CLI. For further information on specific performance data see Section 5.6.3 on page 45.

### 5.3.8 Software Management

The software management of STN includes the following:

- Software inventory: Retrieval of software version information.
- Software upgrade:
  - Download of software package from SMRS to STN.
  - Activation of downloaded software.

Administrative information about SMRS (URL, username and password) is provided by OSS in the STN configuration.

The operator handles administrative information, download, and activation operations with CLI operations (see Reference [21]). The commands are sent over the SFTP protocol, in SSH.

## 5.4 O&M of Baseband Radio Node and Baseband T605

### 5.4.1 General

O&M support for the Baseband Radio Node is implemented in OSS-RC and Baseband Radio Node is seen as a separate network element. The OSS-RC support depends on the OSS-RC revision used.

The Baseband Radio Node is a multi standard node that supports all radio access technologies. This section describes the O&M functionality from a GSM perspective. For more information see Reference [30].

### 5.4.2 Configuration Management

All OSS RC management toward the BSC related to the Baseband Radio Node support is done with the same applications as for GSM DU Radio Nodes. The major applications are CNA for cell configuration and BSM for sector TG management and related attributes. Configuration management for Baseband Radio Node and Baseband T605 is performed via Bulk-CM and Common Explorer

### 5.4.3 Baseband Radio Node Parameter Configuration from BSC

BSC transfers the following Abis over IP related parameters to Baseband Radio Node after connection establishment:

- DiffServ code per traffic type
- Bundling parameters: bundling packet size and bundling time
- Checksum ON or OFF
- Overload Report Threshold
- Overload Report Interval

### 5.4.4 Fault Management

Fault management is performed via Alarm List Viewer (ALV) and Alarm Status Monitoring (ASM).

### 5.4.5 Performance Management

Performance management is performed via PMS.

### 5.4.6 Software Management

Software and hardware management is performed via SHM.

CDM adjust of configuration data is supported for management of Baseband Radio Nodes, not MML adjust.

See the following documents available in OSS RC CPI for further instruction, Reference [34] and Reference [35].

## 5.5 In Service Performance

In BSC, the PGW can get N+1 redundancy by installing an extra RP board. The PGW can also provide redundancy towards the STN and Baseband Radio Nodes by using a floating address as the master IP address (described in Section 6.4 on page 55).

Spare capacity is needed to make the PGW redundancy work. Automatic PGW redundancy is supported.

Redundancy for PGW-RPs is achieved by spare capacity in working RPs. CTH-RP redundancy is achieved by standby RPs.

The above mechanisms are part of the features PGW Load Distribution (for further details see Reference [4]) and CTH Load Distribution (see Reference [5]).

No redundancy is supported for the STN node. Hardware faults in STN may result in loss of all channels handled by that STN. However, the mean time between failure for STN is long enough to not be significant for the in service performance figures.

## 5.6 Related Statistics

### 5.6.1 Abis Statistics for Delay and Packet Loss - STN

The end to end delay for CS and PS frames on Abis is summed up of many delay components. Both CS and PS frames and IP packets can be lost or discarded. See Reference [29] for further information on delays, discarded frames and discarded IP packets.

#### 5.6.1.1 Downlink

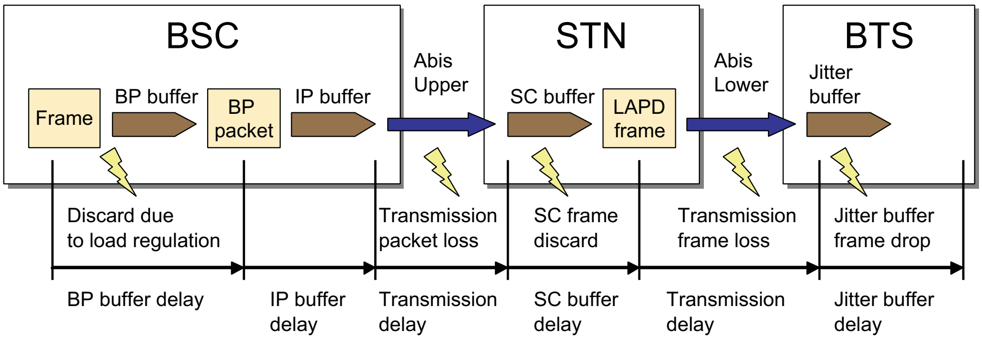


*Figure 7 Delays and Packet Loss Downlink*

This section contains an overview of available statistics for delays and packet loss downlink. Figure 7 shows an overview of points in the network that causes delays and discarded frames and discarded IP packets. Table 4, Table 5 and Table 6 contain information on counters for each separate point in the network together with comments and general information.

*Table 4 Delays Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| BSC BP buffer | - | | Delay according to bundling time for the bundling group (**MCLTDL** in Section 7.1.3 on page 66),<br><br>This statistics counts the average bundling time in bundling groups in downlink. Measured in 1/10 ms.<br>The bundle delay for each bundle is measured as the difference in time between when L2TP packet was sent and when the first frame was entered into the bundling buffer. For each L2TP packet the measured delay is accumulated per bundling profile (LBG) together with number of bundles so that an average delay value can be calculated for each minute. The counter will present an calculated average for the last 15 minutes.<br>The operator can monitor the average delay per traffic type that the bundlingcauses to frames passing the bundling groups in downlink. With this information the operator can then adjust maximum bundling size and maximum bundling time for the bundling groups, to achieve wanted trade off between delay and bandwidth savings. All counters are per TG and per traffic type. |
| | **BUNDG0AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=0 (RSL). |
| | **BUNDG1AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=10 (Speech). |
| | **BUNDG2AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=11 (CS Data). |
| | **BUNDG3AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=12 ((E)GPRS). |
| | **BUNDG4AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=62 (OML). |
| BSC IP buffer | - | | Delay in BSC IP buffers is approximated to 0 ms. |

*Table 4 Delays Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| Abis Upper IP transmission | - | | To get the downlink delay, the round trip is divided by 2. This assumes Abis Upper has a symmetric delay in DL and UL. |
| | **pingDelayAverage**(1) | STN | Average delay of packets received. Measured in 1/10 ms. |
| | **pingPacketCount**(1) | STN | Number of packets received. |
| | **pingDelayCountLT25**(1) | STN | Number of packets received in 0%-25% of pingTimeout. |
| | **pingDelayCountLT50**(1) | STN | Number of packets received in 25% - 50% of pingTimeout. |
| | **pingDelayCountLT75**(1) | STN | Number of packets received in 50% - 75% of pingTimeout. |
| | **pingDelayCountLT100**(1) | STN | Number of packets received in 75% - 100% of pingTimeout. |
| | **pingDelayCountGT100**(1) | STN | Number of packets not received or timed out. |
| | **delayRtAvg**(2) | STN | Average round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtMax**(2) | STN | Maximum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtMin**(2) | STN | Minimum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP95**(2) | STN | 95th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP98**(2) | STN | 98th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP99**(2) | STN | 99th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| STN SC buffer | | | Note that the **AVDELDLSCBUF** counter (located in the BSC SC buffer when the feature Packet Abis over TDM is used) will be zero when Abis over IP is used. |
| Abis Lower transmission | - | | Delay on Abis Lower is approximated to less than 1 ms |

*Table 4 Delays Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| BTS jitter buffer | **DLJITBUFAVDEL** | STS | The Downlink Jitter Buffer delay is measured in the BTS and reported every 5 minutes to the BSC. The counter value is a moving average over the last 15 minutes, calculated from three consecutive values from the BTS. The counter is measured in ms. The statistics is provided per TG and only for CS traffic. |
| | **DL0025JITBUFDEL** **DL2650JITBUFDEL** **DL5175JITBUFDEL** **DL7600JITBUFDEL** **DL100JITBUFDEL** | STS | This statistics shows the delay distribution of CS frames in the Jitter Buffers. The delay distribution is measured in the BTS and reported every 5 minutes to the BSC. The distribution is reported in five intervals uplink and five intervals downlink. The intervals are defined as:<br>0% ≤ delay ≤ 25%<br>26% ≤ delay ≤ 50%<br>51% ≤ delay ≤ 75%<br>76% ≤ delay ≤ 100%<br>delay > 100%<br>The percentage is given relative the size of the Jitter Buffer. The size of the jitter buffer is configurable. The range is estimated for each frame but before a range that represents the actual load is stepped it is rounded to the nearest integer %-value. Statistics is reported per TG and only for CS traffic. See also Section 5.6.1.3 on page 39 |
| | **FJBUFDELDL** | STS | **FJBUFDELDL** counts the accumulated delay [ms] per SC for CS traffic frames in the jitter buffer on the downlink. |
| | **FJBUFDLSCAN** | STS | **FJBUFDLSCAN** counts the number of accumulations for the counter **FJBUFDELDL**. |

*(1) Not supported by Baseband T605*

*(2) Only supported by Baseband T605*

*Table 5 Packet Loss Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| Abis Upper transmission packet loss | **inAbisPacketsErrors** (SIU and TCU) **inAbisPacketErrorsHC** (Baseband T605) | STN | The number of L2TP packets received with checksum errors. |
| | **inAbisPacketsLost** (SIU and TCU) **inAbisPacketLostHC** (Baseband T605) | STN | The number of out-of-sequence or lost L2TP transport session packets received from the BSC. |
| STN SC frame discard | - | | Discarded frames in BTS SC is typicaly 0 or close to 0 due to overprovisioning of the Abis Lower interface. |
| | **SC_FramesLostDownlink** (SIU and TCU) **scFramesLostDownlinkHC** (Baseband T605) | STN | The number of frames discarded by the SC buffer (downlink). Discards are caused by e g temporary overload, or link failure. |
| | **SC_FramesDownlink** (SIU and TCU) **SC_FramesDownlinkHC** (baseband T605) | STN | The total number of frames handled by the SC buffer (downlink), including both discarded and sent. |
| | **TOTFRDLSCBUF** | STS | This counter counts the total number of CS traffic frames entering the super channel buffers downlink, in the PGW, that is frames that are sent to the BTS plus frames discarded by the super channel buffers. |
| | **TOTDLPSSCFRBUF** | STS | This counter counts the total number of PS traffic frames entering the super channel buffers downlink, in the PGW, that is frames that are sent to the BTS plus frames discarded by the super channel buffers. |
| | - | | The **DLCSSCBUFTHR** and **DLPSSCBUFTHR** counters (located in the BSC SC buffer when the feature Packet Abis over TDM is used) will be zero when Abis over IP is used. |

*Table 5 Packet Loss Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| Abis Lower transmission frame loss | - | | An E1/T1 link has a Block Error Ratio (BER) of 1x10-6 or better in normal cases, hence the Abis Lower bit error rate is approximated to 0. |
| BTS jitter buffer discard | **DLDROPJBUF** | STS | This statistics shows the number of frames that has been discarded in the Jitter Buffers due to frames arriving too late to the buffer, frames arriving in the incorrect order or bad content in the TFP header information. The operator can use this statistics to supervise the traffic on the IP links and by that check if the quality and characteristics of the jitter buffers are as expected. The statistics can also be used to trim the size of the Jitter Buffer. The statistics is per TG, uplink/downlink, and only for CS traffic. |
| BTS | **LOSTDLPACK** | STS | Number of lost frames on the DL during last recording period (normally 15 minutes). This includes all CS and PS frames that are missing in the BTS, that is frames that were corrupted in the transmission network as well as frames that were not sent by the PGW due to super channel overload. |
| | **FCSLOSTDL** | STS | **FCSLOSTDL** counts the total number of lost CS traffic frames on the downlink. |
| | **FPSLOSTDL** | STS | **FPSLOSTDL** counts the total number of lost PS traffic frames on the downlink. |

*Table 6 Services Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| BSC | **CSSENTDL** | STS | **CSSENTDL** counts the total number of CS traffic frames sent on the downlink. Reported per TG. |
| | **PSSENTDL** | STS | **PSSENTDL** counts the total number of PS traffic frames sent on the downlink. Reported per TG. |

### 5.6.1.2 Uplink

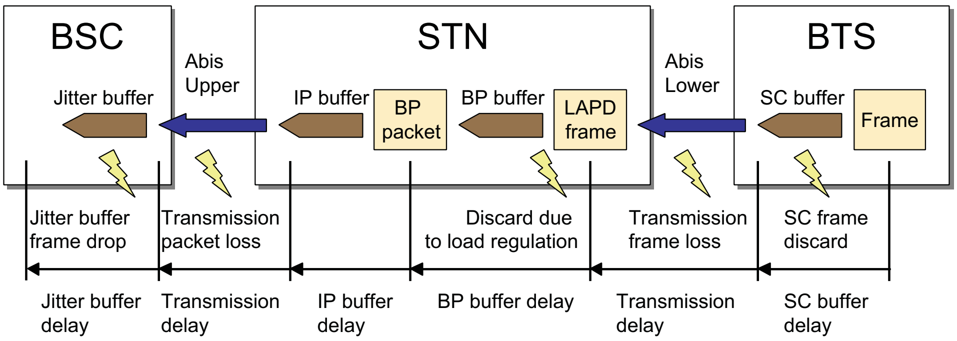


*Figure 8 Delays and Packet Loss Uplink*

This chapter contains an overview of available statistics for delays and packet loss uplink. Figure 8 shows an overview of points in the network that causes delays and discarded frames and discarded IP packets. Table 7 and Table 8 contain information on counters for each separate point in the network together with comments and general information.

*Table 7 Delays Uplink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| BTS SC buffer | **AVDELULSCBUF** | STS | The Uplink Super Channel Buffer delay is measured in the BTS and reported every 5 minutes to the BSC. The counter value is a moving average over the last 15 minutes, calculated from three consecutive values from the BTS. The counter is measured in ms and the average delay is calculated for all frames (CS + PS). |
| | **FSCBUFDELUL** | STS | **FSCBUFDELUL** counts the accumulated delay [ms] for CS traffic frames and PS traffic frames, in the super channel buffer on the uplink. |
| | **FSCBUFULSCAN** | STS | **FSCBUFULSCAN** counts the number of accumulations for the counter **FSCBUFDELUL**. |
| Abis Lower transmission | - | | Delay on Abis Lower is approximated to less than 1 ms |
| STN BP buffer | - | | Delay according to bundling time for the bundling group (**MCLTUL** in Section 7.1.3 on page 66). |
| STN IP buffer | - | | Delay in STN IP buffers is approximated to 0 ms |
| Abis Upper IP transmission | - | | To get the uplink delay, the round trip is divided by 2. This assumes Abis Upper has a symmetric delay in DL and UL.<br>STN of types SIU, TCU and RBS 2409 provides a ping based measurement that can be used. Baseband T605 supports a TWAMP based measurement that can be used. |
| | **pingDelayAverage**(1) | STN | Average delay of packets received. Measured in 1/10 ms. |
| | **pingPacketCount**(1) | STN | Number of packets received. |
| | **pingDelayCountLT25**(1) | STN | Number of packets received in 0%-25% of pingTimeout. |
| | **pingDelayCountLT50**(1) | STN | Number of packets received in 25% - 50% of pingTimeout. |
| | **pingDelayCountLT75**(1) | STN | Number of packets received in 50% - 75% of pingTimeout. |
| | **pingDelayCountLT100**(1) | STN | Number of packets received in 75% - 100% of pingTimeout. |
| | **pingDelayCountGT100**(1) | STN | Number of packets not received or timed out. |
| | **delayRtAvg**(2) | STN | Average round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtMax**(2) | STN | Maximum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtMin**(2) | STN | Minimum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP95**(2) | STN | 95th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP98**(2) | STN | 98th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP99**(2) | STN | 99th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |

*Table 7 Delays Uplink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| BSC jitter buffer | **ULJITBUFAVDEL** | STS | The Uplink Jitter Buffer delay is measured in the BSC and reported every 1 minute. The counter value contains the average value for the last 15 minutes. The counter is measured in ms. The statistics is provided per TG and only for CS traffic. |
| | **UL0025JITBUFDEL**<br>**UL2650JITBUFDEL**<br>**UL5175JITBUFDEL**<br>**UL7600JITBUFDEL**<br>**UL100JITBUFDEL** | STS | This statistics shows the delay distribution of CS frames in the Jitter Buffers. The delay distribution is measured in the BSC and reported every 1 minute. The distribution is reported in five intervals uplink and five intervals downlink. The intervals are defined as:<br>0% ≤ delay ≤ 25%<br>26% ≤ delay ≤ 50%<br>51% ≤ delay ≤ 75%<br>76% ≤ delay ≤ 100%<br>delay > 100%<br>The percentage is given relative the size of the Jitter Buffer. The size of the jitter buffer is configurable. The range is estimated for each frame but before a range that represents the actual load is stepped it is rounded to the nearest integer %-value. Statistics is reported per TG and only for CS traffic. See also Section 5.6.1.3 on page 39. |
| | **FJBUFDELUL** | STS | **FJBUFDELUL** counts the accumulated delay [ms] for CS traffic frames in the jitter buffer on the uplink. |
| | **FJBUFULSCAN** | STS | **FJBUFULSCAN** counts the number of accumulations for the counter **FJBUFDELUL**. |

*(1) Not supported by Baseband T605*

*(2) Only supported by Baseband T605*

*Table 8 Packet Loss Uplink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| BTS SC frame discard | - | | Discarded frames in BTS SC is typically 0 or close to 0 due to overprovisioning of the Abis Lower interface. |
| | **ULSCBUFTHR**<br>**ULPSSCBUFTHR** | STS | This statistics shows the number of frames that has been discarded in the SC buffers due to too many frames in the buffer when emptied. The operator can use this statistics to supervise the traffic on the IP links and by that check if the sizes of the IP links are sufficient. The statistics is divided in PS and CS traffic. |
| | **TOTFRULSCBUF**<br>**TOTULPSSCFRBUF** | STS | This statistics counts total number of frames passing through the SC buffers. The statistics can be used in calculation of for instance ratio of discarded frames. The statistics is separated in CS/PS and UL/DL. |
| | **SC_FramesUplink** (SIU and TCU)<br>**scFramesUplinkHC** (Baseband T605) | STN | The number of frames received by the SC buffer (uplink excluding error frames). |
| Abis Lower transmission frame loss | - | | An E1/T1 link has a Block Error Ratio (BER) of $1x10^{-6}$ or better in normal cases, hence the Abis Lower bit error rate is approximated to 0. |
| Discard in STN due to overload | **uplinkLAPDFramesDiscarded** (SIU and TCU)<br>**uplinkLAPDFramesDiscardedHC** (Baseband T605) | STN | Number of uplink LAPD frames, which had to be discarded, because there was no active Transport Session to carry frames with the relevant SAPI value. This could for example be when the connection to the BSC has been completely or partly lost, or in overload situations when the BSC has throttled down traffic and reconfigured the STN to throw frames. |
| Abis Upper transmission packet loss | **IPLOSTPACKUL** | STS | Number of Abis IP packets being lost on the IP link, reported per TG. |

*Table 8 Packet Loss Uplink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| BSC jitter buffer frame discard | **ULDROPJBUF** | STS | This statistics shows the number of frames that has been discarded in the Jitter Buffers due to frames arriving too late to the buffer, frames arriving in the incorrect order or bad content in the TFP header information. The operator can use this statistics to supervise the traffic on the IP links and by that check if the quality and characteristics of the jitter buffers are as expected. The statistics can also be used to trim the size of the Jitter Buffer. The statistics is per TG, uplink/downlink, and only for CS traffic.<br>Example of how to calculate ratio for discarded CS frames UL:<br>Total Number of CS Frames UL = *UL0025JITBUFDEL+UL2650JITBUFDEL +UL5175JITBUFDEL+UL7600JITBUFDEL+UL100JITBUFDEL*<br>RATIO UL =*(ULDROPJBUF / Total Number of CS Frames UL) x 100* |
| BSC | **LOSTULPACK** | STS | **LOSTULPACK** counts the total number of lost CS traffic frames and PS traffic frames on the uplink. |
| | **FCSLOSTUL** | STS | **FCSLOSTUL** counts the total number of lost CS traffic frames on the uplink, reported per SC. |
| | **FPSLOSTUL** | STS | **FPSLOSTUL** counts the total number of lost PS traffic frames on the uplink, reported per SC. |
| | **CSLOSTUL** | STS | **CSLOSTUL** counts the total number of lost CS traffic frames on the uplink, reported per TG. |
| | **PSLOSTUL** | STS | **PSLOSTUL** counts the total number of lost PS traffic frames on the uplink, reported per TG. |

### 5.6.1.3 Jitter buffer statistics

The jitter buffer shall create a constant flow from a packetized source that contains jitter. In order to do this it needs to store frames. The longer the buffer stores frames the more jitter it can handle.

The configured jitter buffer size is actually the wanted/target average time a frame shall spend in the buffer or in other words the average filling of the jitter buffer. This means that some frames will spend less time in the jitter buffer (these are the frames that were delayed more than average) and some frames will spend more time in the jitter buffer (these are the frames that were a little bit faster than average).

The jitter buffers are monitored in the UL by the counters UL0025JITBUFDEL, UL2650JITBUFDEL, UL5175JITBUFDEL, UL7600JITBUFDEL and UL100JITBUFDEL and in the DL by the counters DL0025JITBUFDEL, DL2650JITBUFDEL, DL5175JITBUFDEL, DL7600JITBUFDEL and DL100JITBUFDEL.

These counters are created mainly to supervise the frames that are delayed in the IP transport, these frames spend less time in the jitter buffer and are tracked with the less than 100% STS counters. It is perfectly normal that the DL100JITBUFDEL is stepping most since this counter are counting all frames that are a little bit faster than average.

For a jitter with a normal distribution DL100JITBUFDEL should have the same value as all the other counters together. Due to the fact that only one counter (DL100JITBUFDEL) is counting the faster than average frames while it exists five counters (DLDROPJBUF, DL0025JITBUFDEL, DL2650JITBUFDEL, DL5175JITBUFDEL, DL7600JITBUFDEL) that are counting slower than average frames. The same applies for the UL counters.

Since jitter is usually more of pareto distributed it is expected that DL100JITBUFDEL have more than half of all the stepping. The second largest counter is expected to be DL7600JITBUFDEL.

On top of this the jitter buffer is adapting to transport delay changes.

There is no UL jitter buffer present in BSC when feature A over IP is active.

### 5.6.2 Abis Statistics for Delay and Packet Loss - Baseband Radio Node

The end to end delay for CS and PS frames on Abis is summed up of many delay components. IP packets can be lost. An IP packet can encapsulate one or several CS and PS frames. See Reference [29] for further information on delays.

#### 5.6.2.1 Downlink

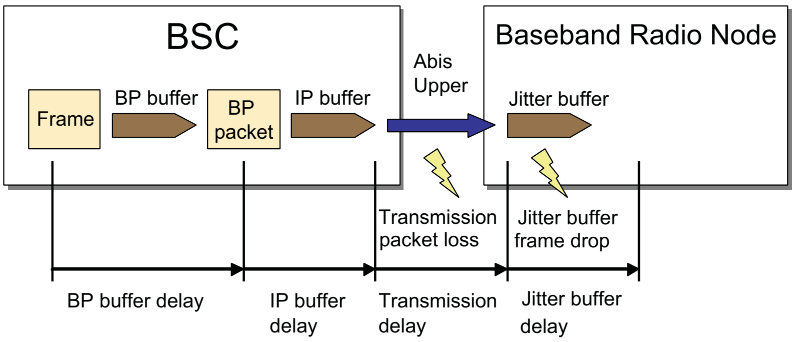


*Figure 9 Delays and Packet Loss Downlink*

This section contains an overview of available statistics for delays and packet loss downlink. Figure 9 shows an overview of points in the network that causes delays and discarded frames and discarded IP packets. Table 9, Table 10 and Table 11 contain information on counters for each separate point in the network together with comments and general information.

*Table 9 Delays Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| BSC BP buffer | - | | Delay according to bundling time for the bundling group (**MCLTDL** in Section 7.1.3 on page 66),<br><br>This statistics counts the average bundling time in bundling groups in downlink. Measured in 1/10 ms.<br>The bundle delay for each bundle is measured as the difference in time between when L2TP packet was sent and when the first frame was entered into the bundling buffer. For each L2TP packet the measured delay is accumulated per bundling profile together with number of bundles so that an average delay value can be calculated for each minute. The counter will present an calculated average for the last 15 minutes.<br>The operator can monitor the average delay per traffic type that the bundling causes to frames passing the bundling groups in downlink. With this information the operator can then adjust maximum bundling size and maximum bundling time for the bundling groups, to achieve wanted trade off between delay and bandwidth savings. All counters are per TG and per traffic type. |
| | **BUNDG0AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=0 (RSL). |
| | **BUNDG1AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=10 (Speech). |
| | **BUNDG2AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=11 (CS Data). |
| | **BUNDG3AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=12 ((E)GPRS). |
| | **BUNDG4AVEDL** | STS | Average bundling delay in the bundling groups containing SAPI=62 (OML). |
| BSC IP buffer | - | | Delay in BSC IP buffers is approximated to 0 ms. |
| Abis Upper IP transmission | - | | To get the downlink delay, the round trip is divided by 2. This assumes Abis Upper has a symmetric delay in DL and UL. |
| | **delayRtAvg** | Baseband Radio Node | Average round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtMax** | Baseband Radio Node | Maximum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtMin** | Baseband Radio Node | Minimum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP95** | Baseband Radio Node | 95th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP98** | Baseband Radio Node | 98th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP99** | Baseband Radio Node | 99th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |

*Table 9    Delays Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| Baseband Radio Node jitter buffer | **pmJitBufDelSamp** | Baseband Radio Node | Number of jitter buffer delay samples. |
| | **pmJitBufDelSum** | Baseband Radio Node | Counts accumulated jitter buffer delay. |
| | **pmJitBufDelSumSq** | Baseband Radio Node | Accumulated squares of jitter buffer delay. |

*Table 10    Packet Loss Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| Abis Upper transmission packet loss | **pmL2tpLostPack** | Baseband Radio Node | Counts sequence gap between two OK packets. Packets out-of-sequence or with wrong CRC are not counted but can indirectly contribute to this counter if considered lost. |
| Baseband Radio Node | **pmCsLostDl** | Baseband Radio Node | **pmCsLostDl** counts lost TFP frames based on sequence number, discarded frames are excluded. |
| | **pmPsLostDl** | Baseband Radio Node | **pmPsLostDl** counts lost P-GSL frames based on sequence number, discarded frames are excluded. |

*Table 11    Services Downlink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| BSC | **CSSENTDL** | STS | **CSSENTDL** counts the total number of CS traffic frames sent on the downlink. Reported per TG. |
| | **PSSENTDL** | STS | **PSSENTDL** counts the total number of PS traffic frames sent on the downlink. Reported per TG. |

### 5.6.2.2 Uplink

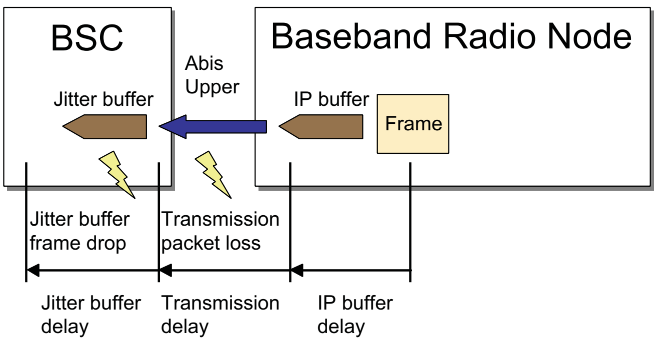


*Figure 10    Delays and Packet Loss Uplink*

This chapter contains an overview of available statistics for delays and packet loss uplink. Figure 10 shows an overview of points in the network that causes delays and discarded frames and discarded IP packets. Table 12, Table 13 and

Table 14 contain information on counters for each separate point in the network together with comments and general information.

*Table 12 Delays Uplink*

| Point in the Network | Counter | Type | Comment |
|---|---|---|---|
| Baseband Radio Node BP buffer | - | Baseband Radio Node | Delay according to bundling time for the bundling group (**MCLTUL** in Section 7.1.3 on page 66). |
| Baseband Radio Node IP buffer | - | Baseband Radio Node | Delay in Baseband Radio Node IP buffers is approximated to 0 ms |
| BSC jitter buffer | **ULJITBUFAVDEL** | STS | The Uplink Jitter Buffer delay is measured in the BSC and reported every 1 minute. The counter value contains the average value for the last 15 minutes. The counter is measured in ms. The statistics is provided per TG and only for CS traffic. |
| | **UL0025JITBUFDEL**<br>**UL2650JITBUFDEL**<br>**UL5175JITBUFDEL**<br>**UL7600JITBUFDEL**<br>**UL100JITBUFDEL** | STS | This statistics shows the delay distribution of CS frames in the Jitter Buffers. The delay distribution is measured in the BSC and reported every 1 minute. The distribution is reported in five intervals uplink and five intervals downlink. The intervals are defined as:<br>0% ≤ delay ≤ 25%<br>26% ≤ delay ≤ 50%<br>51% ≤ delay ≤ 75%<br>76% ≤ delay ≤ 100%<br>delay > 100%<br>The percentage is given relative the size of the Jitter Buffer. The size of the jitter buffer is configurable. The range is estimated for each frame but before a range that represents the actual load is stepped it is rounded to the nearest integer %-value. Statistics is reported per TG and only for CS traffic. See also Section 5.6.2.3 on page 44. |
| | **FJBUFDELUL** | STS | **FJBUFDELUL** counts the accumulated delay [ms] for CS traffic frames in the jitter buffer on the uplink. |
| | **FJBUFULSCAN** | STS | **FJBUFULSCAN** counts the number of accumulations for the counter **FJBUFDELUL**. |
| Abis Upper IP transmission | - | | To get the uplink delay, the round trip is divided by 2. This assumes Abis Upper has a symmetric delay in DL and UL. |
| | **delayRtAvg** | Baseband Radio Node | Average round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtMax** | Baseband Radio Node | Maximum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtMin** | Baseband Radio Node | Minimum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP95** | Baseband Radio Node | 95th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP98** | Baseband Radio Node | 98th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |
| | **delayRtP99** | Baseband Radio Node | 99th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Measured in micro seconds. |

*Table 13 Packet Loss Uplink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| Abis Upper transmission packet loss | **IPLOSTPACKUL** | STS | Number of Abis IP packets being lost on the IP link, reported per TG. |
| BSC jitter buffer frame discard | **ULDROPJBUF** | STS | This statistics shows the number of frames that has been discarded in the Jitter Buffers due to frames arriving too late to the buffer, frames arriving in the incorrect order or bad content in the TFP header information. The operator can use this statistics to supervise the traffic on the IP links and by that check if the quality and characteristics of the jitter buffers are as expected. The statistics can also be used to trim the size of the Jitter Buffer. The statistics is per TG, uplink/downlink, and only for CS traffic.<br>Example of how to calculate ratio for discarded CS frames UL:<br>Total Number of CS Frames UL = *UL0025JITBUFDEL+UL2650JITBUFDEL +UL5175JITBUFDEL+UL7600JITBUFDEL+UL100JITBUFDEL*<br>RATIO UL =*(ULDROPJBUF / Total Number of CS Frames UL) x 100* |
| BSC | **CSLOSTUL** | STS | **CSLOSTUL** counts the total number of lost CS traffic frames on the uplink, reported per TG. |
| | **PSLOSTUL** | STS | **PSLOSTUL** counts the total number of lost PS traffic frames on the uplink, reported per TG. |

*Table 14 Services Uplink*

| Point in the Network | Counter(s) | Type | Comment |
|---|---|---|---|
| Baseband Radio Node | **pmCsSentUl** | Baseband Radio Node | **pmCsSentUl** counts sent TFP frames. Uplink direction only. |
| | **pmPsSentUl** | Baseband Radio Node | **pmPsSentUl** counts sent P-GSL frames. Uplink direction only. |

#### 5.6.2.3 Jitter buffer statistics

The jitter buffer shall create a constant flow from a packetized source that contains jitter. In order to do this it needs to store frames. The longer the buffer stores frames the more jitter it can handle.

The configured jitter buffer size is actually the wanted/target average time a frame shall spend in the buffer or in other words the average filling of the jitter buffer. This means that some frames will spend less time in the jitter buffer (these are the frames that were delayed more than average) and some frames will spend more time in the jitter buffer (these are the frames that were a little bit faster than average).

The jitter buffers are monitored in the UL by the counters UL0025JITBUFDEL, UL2650JITBUFDEL, UL5175JITBUFDEL, UL7600JITBUFDEL and UL100JITBUFDEL and in the DL by the counters pmJitBufDelSamp, pmJitBufDelSum and pmJitBufDelSumSq.

These counters are created mainly to supervise the frames that are delayed in the IP transport, these frames spend less time in the jitter buffer and are tracked with the less than 100% STS counters. It is perfectly normal that the

UL100JITBUFDEL is stepping most since this counter are counting all frames that are a little bit faster than average.

For a jitter with a normal distribution UL100JITBUFDEL should have the same value as all the other counters together. Due to the fact that only one counter (UL100JITBUFDEL) is counting the faster than average frames while it exists five counters (ULDROPJBUF, UL0025JITBUFDEL, UL2650JITBUFDEL, UL5175JITBUFDEL, UL7600JITBUFDEL) that are counting slower than average frames.

Since jitter is usually more of pareto distributed it is expected that UL100JITBUFDEL have more than half of all the stepping. The second largest counter is expected to be UL7600JITBUFDEL.

On top of this the jitter buffer is adapting to transport delay changes.

There is no UL jitter buffer present in BSC when feature A over IP is active.

### 5.6.3 Monitoring of BSC - STN Link

The following counters are used to monitor Abis when Abis over IP is used. More information on how to use the counters can be found in Reference [8]. The operator can monitor the throughput on the IP link between BSC and STN in order to determine if the correct dimensioning applies.

The following STS counters are available from BSC:

*Table 15 Counters for monitoring BSC — STN link*

| Name | Type | Description |
|---|---|---|
| **IPSENTKBYTES** | STS | Accumulated number of kilo bytes of DL data sent by the BSC, reported per TG. The measurement includes all Speech, CS data, GPRS, RSL signaling, OML signaling and L2TP Control Connection. The measurement includes the length of the entire IP packets including IP header. |
| **IPRECKBYTES** | STS | Accumulated number of kilo bytes of UL data received by the BSC, reported per TG. The measurement includes all Speech, CS data, GPRS, RSL signaling, OML signaling and L2TP Control Connection. The measurement includes the length of the entire IP packets including IP header. |
| **IPLOSTPACKUL** | STS | Number of Abis IP packets being lost on the IP link, reported per TG. |
| **IPNUMSCAN** | STS | Number of scans of the IPSENTKBYTES and IPRECKBYTES counters, reported per TG. The counters are scanned each second. This counter will represent the time in seconds for which the counters have been accumulated. |
| **IPULRECPACK** | STS | Number of Abis IP packets received by the BSC, reported per TG. |
| **IPDLSENTPACK** | STS | Number of Abis IP packets sent by the BSC, reported per TG. |
| **IPOVLPSREG** | STS | Indicates how long time the PS traffic reduction has been active for Abis over IP. Measured in seconds. Stepping of this counter indicates that a number of PS data scheduling has been omitted. |

*Table 15 Counters for monitoring BSC — STN link*

| Name | Type | Description |
|---|---|---|
| **IPOVLCSREG** | STS | Indicates how long time the CS traffic reduction has been active for Abis over IP. Measured in seconds. Stepping of this counter indicates that as good as all PS data scheduling has been omitted. Stepping of this counter also indicates a decreased level of new and existing CS calls. |
| **UL7075STNLOAD**<br>**UL7680STNOAD**<br>**UL8185STNLOAD**<br>**UL8690STNLOAD**<br>**UL9195STNLOAD**<br>**UL100STNLOAD**<br>**DL9600STNLOAD**<br>**DL7075STNLOAD**<br>**DL7680STNLOAD**<br>**DL8185STNLOAD**<br>**DL8690STNLOAD**<br>**DL9195STNLOAD**<br>**DL9600STNLOAD**<br>**DL100STNLOAD** | STS | Load per STN in both uplink and downlink is monitored with histogram counters. The counters counts time above a specific load percentage per TG. TG counters connected to one STN are accumulated. Since the load is calculated against an engineered bandwidth and not the real bandwidth it is possible to have a load that exceeds 100%. Also note that it is possible for an STN to handle TGs connected to more than one BSC. If that is the case, these counters will not reflect the total bandwidth need to that STN, only TGs handled in this BSC are included.<br>The counters are scanned every 1s and a counter with a range that represents the actual load is stepped.<br>Formula to calculate DLxxxxSTNLOAD:<br>*(sum of bits sent per second via all TGs connected to the STN/sum of maximal bandwidth of all TGs connected to the STN)*100%* |

The following counters are available from STN:

*Table 16 Counters to monitor the BSC-STN link*

| Name | Type | Description |
|---|---|---|
| **inAbisOctets** (SIU and TCU)<br>**inAbisOctetsHC** (Baseband T605) | STN | The number of octets received from the BSC. The counter counts transport session packets. Control connection messages are not included. The counter counts the load contributed by transport session packets at the IP level. Each packet increments the counter with the length of the entire IP packet, including the IP header. |
| **outAbisOctets** (SIU and TCU)<br>**outAbisOctetsHC** (Baseband T605) | STN | The number of octets sent to the BSC. The counter counts transport session packets. Control connection messages are not included. The counter counts the load contributed by transport session packets at the IP level. Each packet increments the counter with the length of the entire IP packet, including the IP header. |
| **inAbisPackets** (SIU and TCU)<br>**inAbisPacketsHC** (Baseband T605) | STN | The number of L2TP transport session packets received from the BSC, excluding control connection messages. |
| **outAbisPackets** (SIU and TCU)<br>**outAbisPacketsHC** (Baseband T605) | STN | The number of L2TP transport session packets sent to the BSC, excluding control connection messages. |
| **inAbisPacketsErrors** (SIU and TCU)<br>**inAbisPacketsErrorsHC** (Baseband T605) | STN | The number of L2TP packets received with checksum errors. |
| **inAbisPacketsLost** (SIU and TCU)<br>**inAbisPacketsLostHC** (Baseband T605) | STN | The number of out-of-sequence or lost L2TP transport session packets received from the BSC. |

### 5.6.4 Monitoring of BSC - Baseband Radio Node Link

The following counters are used to monitor Abis when Abis over IP is used. More information on how to use the counters can be found in Reference [8]. The operator can monitor the throughput on the IP link between BSC and Baseband Radio Node in order to determine if the correct dimensioning applies.

The following STS counters are available from BSC:

*Table 17 Counters for monitoring BSC — Baseband Radio Node link*

| Name | Type | Description |
|---|---|---|
| **IPSENTKBYTES** | STS | Accumulated number of kilo bytes of DL data sent by the BSC, reported per TG. The measurement includes all Speech, CS data, GPRS, RSL signaling, OML signaling and L2TP Control Connection. The measurement includes the length of the entire IP packets including IP header. |
| **IPRECKBYTES** | STS | Accumulated number of kilo bytes of UL data received by the BSC, reported per TG. The measurement includes all Speech, CS data, GPRS, RSL signaling, OML signaling and L2TP Control Connection. The measurement includes the length of the entire IP packets including IP header. |
| **IPLOSTPACKUL** | STS | Number of Abis IP packets being lost on the IP link, reported per TG. |
| **IPNUMSCAN** | STS | Number of scans of the IPSENTKBYTES and IPRECKBYTES counters, reported per TG. The counters are scanned each second. This counter will represent the time in seconds for which the counters have been accumulated. |
| **IPULRECPACK** | STS | Number of Abis IP packets received by the BSC, reported per TG. |
| **IPDLSENTPACK** | STS | Number of Abis IP packets sent by the BSC, reported per TG. |
| **IPOVLPSREG** | STS | Indicates how long time the PS traffic reduction has been active for Abis over IP. Measured in seconds. Stepping of this counter indicates that a number of PS data scheduling has been omitted. |
| **IPOVLCSREG** | STS | Indicates how long time the CS traffic reduction has been active for Abis over IP. Measured in seconds. Stepping of this counter indicates that as good as all PS data scheduling has been omitted. Stepping of this counter also indicates a decreased level of new and existing CS calls. |

The following counters are available from Baseband Radio Node:

*Table 18 Counters to monitor the BSC-Baseband Radio Node link*

| Name | Type | Description |
|---|---|---|
| **pmL2tpRecPack** | Baseband Radio Node | The number of L2TP packets received from the BSC. Packets out-of-sequence or with wrong CRC are not counted. Duplicated counts as out-of-sequence. |
| **pmL2tpSentPack** | Baseband Radio Node | The number of L2TP packets sent to the BSC. |
| **pmL2tpLostPack** | Baseband Radio Node | Counts lost L2TP packets in the downlink direction.. Packets out-of-sequence or with wrong CRC are not counted |

### 5.6.5 TWAMP Measurements

If TWAMP measurements are active in Baseband T605 or Baseband Radio Node the following counters can be used to get detailed information on the Abis IP link characteristics. See also Section 6.10 on page 60

*Table 19 Counters to monitor the Abis IP link using TWAMP*

| Name | Description |
|---|---|
| **delayRtAvg** | Average round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Unit: 1 microsecond |
| **delayRtMax** | Maximum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Unit: 1 microsecond |

| | |
|---|---|
| **delayRtMin** | Minimum round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Unit: 1 microsecond |
| **delayRtP95** | 95th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Unit: 1 microsecond |
| **delayRtP98** | 98th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Unit: 1 microsecond |
| **delayRtP99** | 99th percentile value of round trip delay between TWAMP initiator and TWAMP responder, where Round trip delay is (T4-T1)-(T3-T2). Unit: 1 microsecond |
| **dscpRecMax** | Maximum received Differentiated Services Code Point (DSCP) value. |
| **dscpRecMin** | Minimum received Differentiated Services Code Point (DSCP) value. |
| **duplicPktFwd** | Duplicated packets in initiator to responder direction.. |
| **duplicPktRev** | Duplicated packets in responder to initiator direction. |
| **ipdvFwdAvg** | Average Inter Packet Delay Variation (IPDV) measured in initiator to responder direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvFwdMax** | Maximum IPDV measured in initiator to responder direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvFwdMin** | Minimum IPDV measured in initiator to responder direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvFwdP95** | 95th percentile value of IPDV measured in initiator to responder direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvFwdP98** | 98th percentile value of IPDV measured in initiator to responder direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvFwdP99** | 99th percentile value of IPDV measured in initiator to responder direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvRevAvg** | Average IPDV measured in responder to initiator direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvRevMax** | Maximum IPDV measured in responder to initiator direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvRevMin** | Minimum IPDV measured in responder to initiator direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvRevP95** | 95th percentile value of IPDV measured in responder to initiator direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **ipdvRevP98** | 98th percentile value of IPDV measured in responder to initiator direction. Only zero to positive values is presented. Unit: 1 microsecond |

| | |
|---|---|
| **ipdvRevP99** | 99th percentile value of IPDV measured in responder to initiator direction. Only zero to positive values is presented. Unit: 1 microsecond |
| **lostPeriodMaxFwd** | Maximum lost period in millisecond in initiator to responder direction. Example: Three consecutive packet is lost when frequency is 10pps=300ms. |
| **lostPeriodMaxRev** | Maximum lost period in milliseconds in responder to initiator direction. Example: Three consecutive packet is lost when frequency is 10pps=300ms. |
| **lostPeriodMinFwd** | Minimum lost period in millisecond in initiator to responder direction. Example: One consecutive packet is lost when frequency is 10pps=100ms. |
| **lostPeriodMinRev** | Minimum lost period in milliseconds in responder to initiator direction. Example: One consecutive packet is lost when frequency is 10pps=100ms. |
| **lostPeriodsFwd** | Number of occasions with lost packets in initiator to responder direction. |
| **lostPeriodsRev** | Number of periods with lost packets in responder to initiator direction. |
| **lostPktsFwd** | Number of lost packets in initiator to responder direction. Packet is considered lost after 4 second. |
| **lostPktsRev** | Number of lost packets in responder to initiator direction. Packet is considered lost after 4 second. |
| **reorderPktFwd** | Reordered packets in initiator to responder direction. |
| **reorderPktRev** | Reordered packets in responder to initiator direction. |
| **rxPkts** | Number of packets received by TWAMP initiator including duplicated packets. |
| **ttlMaxFwd** | Maximum Time To Live (TTL) for IPv4 or Hop Limit for IPv6 value in initiator to responder direction. |
| **ttlMaxRev** | Maximum TTL for IPv4 or Hop Limit for IPv6 value in responder to initiator direction. |
| **ttlMinFwd** | Minimum TTL for IPv4 or Hop Limit for IPv6 value in initiator to responder direction. |
| **ttlMinRev** | Minimum TTL for IPv4 or Hop Limit for IPv6 value in responder to initiator direction |
| **txPkts** | Number of packets sent from TWAMP initiator. |

### 5.6.6 Monitoring of TimeServer - STN Link

The Performance Management (PM) parameters below are only available for STNs of type SIU, TCU and RBS2409.

The measurements described below are supported by reports in OSS.

The operator can monitor the quality of network to each TimeServer. Only TimeServer instances that has been used during the measurement period will contain valid counters. Measurement based on the synchronization evaluation period that differs between synchronizing at cold restart, warm restart and synchronized state exists.

The following Performance Management (PM) parameters are defined and used to monitor the performance of the TimeServer - STN link:

*Table 20 PM parameters to monitor TimeServer - STN link*

| Name | Type | Description |
|---|---|---|
| **TS_NoPkts** | STN | The number of received NTP packets. |
| **TS_TimeOffset_1** | STN | The Time Offset calculated on the best 1% of the received NTP packets. |
| **TS_TimeOffset_10** | STN | The Time Offset calculated on the best 10% of the received NTP packets. |
| **TS_TimeOffset_50** | STN | The Time Offset calculated on the best 50% of the received NTP packets. |
| **TS_RoundTripDelay_50** | STN | The round trip delay calculated on the best 50% of the received NTP packets |
| **TS_NoTS_Reslections** | STN | The number of performed successful TimeServer reselections. |

The best NTP packets for Time Offset calculation are the packets which indicate smallest deviation from the Time Offset of the packet with the smallest round trip delay.

The best NTP packets for round trip delay calculation are the packets with the smallest round trip delay.

The absolute value of the Time Offset has no significance but the variation of the Time Offset. The Time Offset variation is due to following main contributions:

- Variation due to short term packet delay variation on the uplink and downlink paths.
- Variation due to asymmetric packet delay change on the uplink and downlink paths.
- Stepwise path delay changes due to rerouting in the IP network.
- Slow Time Offset change due to frequency difference between the TimeServer and the STN.

The STN calibration algorithm deduces the frequency difference between the TimeServer and the STN from the slow Time Offset change and re-calibrates the high-stable oscillator in the STN.

### 5.6.7 Performance Monitoring of STN and Baseband Radio Node

The counters available in the STN or Baseband Radio Node performance data reported to OSS from each STN or Baseband Radio Node are described in CPI for relevant STN or Baseband Radio Node.

### 5.6.8 Impact on Other Counters

Depending on dimensioning of Abis, the following STS counters may be impacted:

- *Cell congestion counters*

  In case of continuous Abis overload, new connections in the cell will be limited. This will result in increased value of cell congestion counters if the bandwidth on Abis is limited.

- *Abis congestion counters*

  OVERLOADREJCON counts number of rejected new CS connections due to Abis overload per cell. This counter is stepped when an attempt to allocate an idle TCH fails and Abis overload is indicated in the cell. The operator can use this counter to see if there has been overload due to Abis.

  PREJABISCONG counts the number of UL Temporary Block Flows (TBFs) for new MSs (not in DTM) that have been rejected on PDCH:s carried on traffic session that have Abis congestion, for Packet Abis.

  DTMULABISFAILRES counts the number of 44.060 PACKET ACCESS REJECT and 44.018 DTM REJECT sent to the MS during TBF establishment for the reason “Abis Overload”, when the MS is in Dedicated mode or DTM.

The Abis counter values for throughput, delay, jitter and dropped packets are influenced by other services utilizing the same links as Abis. The jitter and delay may for example increase if adding other services. Throughput can decrease and dropped packets can increase if adding a service with higher priority.

# 6 Engineering Guidelines

## 6.1 Hardware

### 6.1.1 BSC

**PGW**

The amount of traffic one PGW-RP or CTH-RP can handle depends on network configuration and transmission load. For details on PGW dimensioning see Reference [11].

Spare capacity is needed to make the PGW redundancy work. Automatic PGW redundancy is supported.

BSC tries to relocate the TGs handled by an RP that fails. BSC also tries to relocate TGs when an RP is experiencing high load. Automatic relocations only take place if suitable target RPs can be found.

Redundancy for PGW-RPs is achieved by spare capacity in working RPs. CTH-RP redundancy is achieved by standby RPs.

The above mechanisms are part of the features PGW Load Distribution (for further details see Reference [4]) and CTH Load Distribution (see Reference [5]).

**BSC IP Network Interfaces**

The BSC LAN Switch and the BSC NWI-E support up to 4096 TRXs.

### 6.1.2 STN

In the Macro RBS case, the STN is realized with the SIU or TCU hardware. One SIU or TCU has the capacity to serve 6 TGs of up to 36 TRXs regardless of traffic model. Up to 72 TRXs can be handled but not for maximum traffic at the same time for all TRXs. The STN, realized by the Baseband T605 hardware has the capacity to serve 6 TGs and up to 72 TRXs. In the RBS 2409, the STN is integrated with the BTS HW, and serves one TG of 1-2 TRXs. No STN redundancy is supported. The hardware capabilities are summarized in Table 1.

### 6.1.3 Baseband Radio Node

Baseband 5212 has the capacity of 24 TRXs in GSM single standard mode. Baseband 5216 has the capacity of 48 TRXs in GSM single standard mode and 24 TRXs in mixed mode baseband. For more information, see Reference [30].

### 6.1.4 TimeServer

The TimeServer supplies time stamps of sufficient accuracy to synchronize STNs and Baseband Radio Nodes. To support redundancy, multiple TimeServers are made available for each node. A node can be connected to a number of TimeServers, but only one connection is active at a time. TimeServer failure results in TimeServer re-selection for all nodes connected to the failing TimeServer. Such an event does not cause the node to lose synchronization. For further information see Reference [7] and Reference [32].

### 6.1.5 SEGW

The Security Gateway (SEGW), located at the BSC site, provides a secure IPSec tunnel towards the RBS site through an external, unsecure transport network. SEGW redundancy is recommended. In this way single points of failure in the SEGW and the connected links can be eliminated.

### 6.1.6 OSS

The Bi-directional SMRS software repository, based on NESS and NEDSS servers, is required. For further information see Reference [13].

### 6.1.7 RBS

See the GSM/BTS Software release note of the RBS versions used.

## 6.2 Requirements on IP Network Characteristics

The requirements on IP network characteristics are described in Reference [29].

## 6.3 Special Adjustments for Adaptive P-GSL Timing Advance

The feature Adaptive Timers for Packet Abis (see **PAL** in Section 7.1.6 on page 74) will automatically adjust the internal PTA value according to the channel delay. To adjust the internal PTA value the BTS measures the round trip time between the BTS and the BSC using ongoing packet switched data traffic. These measurements are used to make an estimate of the round trip time which in turn is utilized in order to update the Packet Timing Advance. By choosing the Packet Timing Advance properly it is ensured that data packets arrive at the transmitter at the correct time in order to be sent over the air interface. Although Adaptive Timers for Packet Abis will adapt the internal PTA value according to the delay of the channel, the delay variation must be taken into account because the regulation does not cope with jitter.

When using internal PTA regulation together with the Abis over IP feature it is important to be aware of the delay variation induced by the setting of the bundling time. A larger bundling time will increase the delay variations thus requiring a larger setting of the **JBPTA** parameter, and a smaller bundling time will decrease the delay variations thus enabling a smaller setting of the **JBPTA** parameter.

If the delay variation is larger than the corresponding value of the **JBPTA** parameter this will lead to an increased packet loss ratio and possibly disconnections.

If the Interfaces over satellite feature is used the **JBPTA** should be set to 60 ms, thus providing for a larger variation in delay.

**Note:** If the feature Reduced Latency (see Reference [25]) is used it is important not to set the **JBPTA** parameter to a higher value than 30 ms, since it then is not possible to use more than half of the TRX buffer depth.

If the delay characteristics of the network are not known or it is desired to minimize the **JBPTA** parameter, the size of the jitter buffer can be tuned in the following manner: set the **JBPTA** parameter according to the recommendation from Reference [10]. Verify that the percentage of lost PS frames downlink (see Reference [8]) is at an acceptable level. Decrease the current setting of the **JBPTA** parameter and verify that there is not any significant increase in the percentage of frames lost on the downlink. Repeat the process until a minimum value of **JBPTA** is found such that there is no significant increase in the percentage of lost PS frames.

**Note:** The delay variation will also be affected by the internal system delay variation caused by high load. If the tuning is performed during a low load scenario it is recommended to be restrictive with the acceptable percentage of lost frames.

## 6.4 IP Addressing

There are several types of IP traffic terminated in the BSC. These traffic types are separated into several subnets, one per traffic type. One of the subnets is the Abis over IP network. A dedicated subnet-based VLAN is configured in the BSC IP network interfaces to carry the Abis over IP traffic to and from the STN or Baseband Radio Node. A public subnet with one IP address per RP, running Abis over IP, one IP address common for all RPs, and one IP address for each of the two BSC IP network interfaces are required. It is recommended to allocate a subnet sufficiently large to cover the largest BSC configuration expected. All traffic to the switches will be tagged internally by the BSC IP network interfaces, based on protocol type or IP subnet, and packets to hosts within BSC will be untagged.

The logical connection towards the site routers is called SR_Abis in the BSC IP network interfaces. It requires one IP address per BSC IP network interfaces. One extra IP address may be required if VRRP is used for site integration. All payload and signaling traffic in and out from BSC will use the same physical port on the BSC IP network interfaces. These external ports (north bound interface) support VRRP and VLAN tagging towards the site routers.

**Note:** Except from where external VLAN tagging is enabled in the configuration, all VLAN names and VLAN tags are internal to the BSC IP network interfaces.

To achieve redundancy for the PGW, one IP address shall be setup as reconfigurable and all STNs and Baseband Radio Nodes shall be configured to use this address when starting to establish the control connection towards BSC. In this way, the STNs and Baseband Radio Nodes will always be able to reach BSC as long as at least one PGW/CTH-RP is up and running.

For more information on IP addressing for the BSC IP applications see Reference [12].

For more information on how to connect the BSC IP network interfaces to the Site Network, see Reference [26] and Reference [27].

## 6.5 Abis Path

When migrating a TG from a classic TDM configuration (without Flexible Abis, LAPD Concentration and Multiplexing), the Abis path definitions simply replace the number of current transmission devices with PGW devices instead, one for one. This creates internal backwards compatibility to the TRH, TRA and GPH. The transmission side of the PGW is also PGW devices but defined within SC definitions (the PGW is inserted in between in the Abis path).

When migrating a TG using Flexible Abis, LAPD Concentration or Multiplexing to Abis over IP, the amount of Abis paths needed to be defined must again be one to one with the channels defined. Each E-TCH or G-TCHs needs its own device, not pooled as for Flexible Abis. Also each TRX needs one device for signalling. Sharing one 64k device as for LAPD Concentration and Multiplexing is not allowed. Thus, the number of the Abis path DCPs will increase.

**Note:** This is not an increase in physical Abis transmission, it is how the BSC internal device definition is required to be.

## 6.6 Quality of Service and Bundling Groups

### 6.6.1 General

IP networks with limited bandwidth and a risk of overload may improve the performance with Quality of Service, if DiffServ is supported by the IP network. In this case, it is recommended to give the RSL signalling, OML signalling, L2TP Control protocol and NTP a higher priority than other transmission, to not risk losing a cell at overload.

**Note:** It is not recommended to try to deploy a base station on a too small bandwidth.

If DiffServ is used, its DSCP value is set or changed by command. What DSCP value to use depends on the IP service used for each BTS site. The principle is to configure the highest priority for signaling (SAPI 0 and 62), to not risk cell outage at overload, and to give CS speech and CS data (SAPI 10 and 11) higher priority than PS data (SAPI 12). On the downlink, L2TP control connections use the DSCP value configured for SAPI 62. In STN and Baseband Radio Node, it is also possible to define DSCP values for Synchronization (NTP) and L2TP control connections (uplink). If Abis over IP is used in combination with transport sharing, the recommendations are further described in Reference [17], Reference [28] and Reference [33].

The bundling group parameters are described in Section 7.1.3 on page 66 and the Load Regulation parameters are described in Section 7.1.5 on page 71.

### 6.6.2 BSC - STN

To configure which traffic types to be bundled together in IP packets on Abis Upper, the operator first needs to define bundling profiles. Secondly, the traffic types are assigned to a bundling profile.

Initially one LAPD bundling profile (LBG) is created (using the default settings shown below):

- **LBG** = 0
- **MCLTUL** = **MCLTDL** = 1 ms
- **MPLSUL** = **MPLSDL** = 1465 bytes
- **DSCPUL** = **DSCPDL** = 0
- **CRC** = ON

LBG 0 is the default bundling profile used if no other bundling profiles are defined. LBG 0 is not possible to change or remove.

When configuring bundling, consider the following rules:

- All traffic types assigned to a specific bundling profile will be bundled together.
- **CRC32** can be set to ON/OFF for each bundling profile.
- The DiffServ value, **DSCPUL** and **DSCPDL**, can be configured independently downlink and uplink for each bundling profile.
- The maximum bundling time, **MCLTDL** and **MCLTUL**, can be configured independently downlink and uplink for each bundling profile.

- The maximum bundling size, **MPLSDL** and **MPLSUL**, can be configured independently downlink and uplink for each bundling profile.

### 6.6.3 BSC - Baseband Radio Node

Traffic types are bundled together in IP packets on Abis using the default settings shown below:

- **MCLTUL** = **MCLTDL** = 1 ms
- **MPLSUL** = **MPLSDL** = 1465 bytes
- **DSCPUL** = **DSCPDL** = 51 for SAPI 0 and 62
- **DSCPUL** = **DSCPDL** = 46 for SAPI 10 and 11
- **DSCPUL** = **DSCPDL** = 28 for SAPI 12
- **CRC** = ON

Theses values can be changed by command.

When configuring bundling, consider the following rules:

- **CRC32** can be set to ON/OFF for each traffic type.
- The DiffServ value, **DSCPUL** and **DSCPDL**, can be configured independently downlink and uplink for each traffic type.
- The maximum bundling time, **MCLTDL** and **MCLTUL**, can be configured independently downlink and uplink for each traffic type.
- The maximum bundling size, **MPLSDL** and **MPLSUL**, can be configured independently downlink and uplink for each traffic type.
- Traffic types that have the same bundling configuration forms a bundling profile. A bundling profile may be shared by a number of bundling groups. Traffic frames in a bundling group are bundled together in IP packets on Abis. The exact number of bundling groups depends on the number of profiles and on the resource allocation and HW limitation factors of the Baseband Radio Node.

## 6.7 Super Channels

Super Channels are used on Abis Lower. Note that for a Baseband Radio Node there is no Abis Lower and therefore no Super Channels. In case of overload on Abis Lower, signalling and CS speech and CS data are prioritized over PS data. This means that the end-user will in most situations only experience a lower packet data throughput at traffic peaks. Unless Abis Lower is substantially under dimensioned there will not be any impact on dropped call rate or speech quality.

1 - 4 Super Channels per TG can be configured on Abis Lower. As a rule of thumb, one Super Channel is enough for 12 TRXs if less than 20 EPDCHs are used. If using two Super Channels with 6 TRXs configured on each Super Channel then 40 - 45 EPDCHs can be configured without risk for decreased packet data throughput at peak. That is in the unlikely situation where all EPDCHs happens to be allocated on one of the Super Channel. Normally the EPDCHs are spread between the two Super Channels which means that more EPDCHs can be used without risk of decreased packed data throughput.

Monitor SC_FramesLostDownlink (ref Section 5.6.1 on page 32) in order to detect if frames are lost due to under dimensioning of Abis Lower. Configuring four Super Channels per TG will cater for all possible traffic scenarios, but two Super Channels are enough for all but the most extreme scenarios and end-user impact is minimized by the prioritizing of traffic on Abis Lower.

## 6.8 SEGW

Security Gateway (SEGW) redundancy is recommended at the BSC, OSS and TimeServer site (in case the operator decides to implement the common network approach then there will be only BSC sites). In this way we can eliminate single points of failure in the SEGW and the connected links.

RBS site redundancy is not assumed.

The STN and Baseband Radio Node have built in support for IPSec VPN.

Ericsson portfolio considers the Juniper (Netscreen) Products as security devices suitable for the Abis over IP Security Solution implementation. Two Juniper devices configured in HA active/passive mode are recommended at the BSC site working as RAN SEGWs. The state of the active gateway, such as connection table and other vital information, is continuously copied to the inactive gateway. When the cluster fails over to the inactive gateway, it knows which connections are active, and communication traffic can continue to flow uninterrupted. Both SEGWs share a number of virtual IP and MAC addresses which are seized by the active SEGW and provide single gateway addresses to SEGWs in RBS sites.

The reference Abis over IP solution is based on the use of route based VPNs between the SEGW at the BSC site and each RBS SEGW. Encapsulating Security Payload (ESP) is used for encryption and authentication. In addition to the routing configuration, policies between the different security zones are needed for each traffic flow.

All basic and HA specific configuration must be done initially at site without support from OSS. HA specific configuration has been performed without OSS support and will not be visible in OSS. OSS assumes that the interfaces on the node/device are symmetrically configured, for example when creating a new VPN tunnel, it shall be possible to bind the VPN tunnel to the same tunnel interface independent of currently active node/device. A SEGW plug-in is described in NCM Support for Security Gateway Nodes (see Reference [13]).

OSS receives and supervises alarms via SNMP from each individual SEGW. OSS assumes that the source IP address of a received SNMP packet from a node contain the logical managed IP address assigned to that node.

It is important to understand that in order to avoid the possibility of accessing the BSC O&M application from the RBS Site by STN IP spoofing, policies between the unsecure zone and the O&M security zone should be carefully configured to allow the proper protocols towards each of the IP destinations (OSS-RC and BSC O&M Applications).

IP overhead using IPSec will affect the dimensioning of the network.

## 6.9 PPPoE

Care should be taken when configuring username and password in the STN part of RBS 2409 for PPPoE. If the same pair of username and password is shared between several RBS 2409, only the last established PPP session, between these RBS 2409 and the BRAS, will remain active. The other PPP sessions, using the same pair of username and password, will be closed. It is therefore necessary to use the username and password specified by the DSL access network provider for each PPPoE connection.

## 6.10 Two Way Active Measurement Protocol

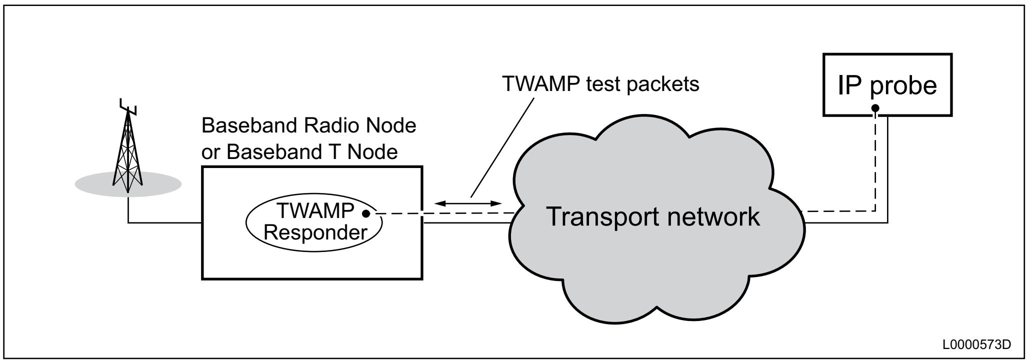


*Figure 11 Measured Network Path*

In Baseband T605 and Baseband Radio Node there is support for Two Way Active Measurement Protocol (TWAMP) measurements of IP network characteristics like round trip time, jitter and loss. These nodes support both TWAMP initiator and TWAMP responder functionality.

For TWAMP to work there need to be both an initiator and a responder in the network, see Figure 11.

At the BTS site the Baseband Radio Node or the Baseband T605 will act as initiator. For configuration details see Page 90 or Page 92.

At the BSC site TWAMP responder functionality must be available. This can be achieved by a number of commercially available products or by placing a Baseband T605 close to the BSC.

TWAMP is described in RFC 5357.

# 7 Parameters

## 7.1 Main Controlling Parameters in BSC

### 7.1.1 BSC Parameter Index

*Table 21 BSC Parameters in Alphabetic Order*

| Parameter Name | Expansion | Subsection |
|---|---|---|
| **APL** | Application identifier | Section 7.1.2 on page 64 |
| **APPLSEL** | Application selected | Section 7.1.2 on page 64 |
| **ATHABIS** | Dynamic half rate allocation Abis | Section 7.1.5 on page 71 |
| **CRC32** | Cyclic redundancy check 32 bits | Section 7.1.3.1 on page 66 and Section 7.1.4 on page 69 |
| **DAMRCRABIS** | Dynamic AMR codec reduction Abis | Section 7.1.5 on page 71 |
| **DCP** | Digital connection point | Section 7.1.3.2 on page 67 and Section 7.1.3.3 on page 69 |
| **DEVNO** | Device number | Section 7.1.2 on page 64 |
| **DSCPDL** | Differential service code point downlink | Section 7.1.3.1 on page 66 and Section 7.1.4 on page 69 |
| **DSCPUL** | Differential service code point uplink | Section 7.1.3.1 on page 66 and Section 7.1.4 on page 69 |
| **IPADDR** | IP addressing | Section 7.1.2 on page 64 |
| **IPDEVNO** | IP device number | Section 7.1.2 on page 64 |
| **IPDEVTYPE** | IP device type | Section 7.1.2 on page 64 |
| **IPOV** | IP overload threshold based on packet loss | Section 7.1.5 on page 71 |
| **IWDVER** | BSC-STN inter work description version | Section 7.1.3.1 on page 66 and Section 7.1.4 on page 69 |
| **JBPTA** | Jitter buffer P-GSL timing advance downlink | Section 7.1.6 on page 74 |
| **JBSDL** | Jitter buffer size downlink | Section 7.1.3.2 on page 67 |
| **JBSUL** | Jitter buffer size uplink | Section 7.1.3.2 on page 67 |
| **LBG** | LAPD bundling group profile | Section 7.1.3.1 on page 66<br>Section 7.1.3.2 on page 67 |
| **LDEL** | LAPD signalling delay | Section 7.1.5 on page 71 |
| **MASK** | IP sub-net mask | Section 7.1.2 on page 64 |
| **MBWDL** | Maximum bandwidth downlink | Section 7.1.3.2 on page 67 and Section 7.1.4 on page 69 |
| **MBWUL** | Maximum bandwidth uplink | Section 7.1.3.2 on page 67 and Section 7.1.4 on page 69 |
| **MCLTDL** | Maximum collection time downlink | Section 7.1.3.1 on page 66 and Section 7.1.4 on page 69 |

| Parameter Name | Expansion | Subsection |
|---|---|---|
| **MCLTUL** | Maximum collection time uplink | Section 7.1.3.1 on page 66 and Section 7.1.4 on page 69 |
| **MPLSDL** | Maximum packet size downlink | Section 7.1.3.1 on page 66 and Section 7.1.4 on page 69 |
| **MPLSUL** | Maximum packet size uplink | Section 7.1.3.1 on page 66 and Section 7.1.4 on page 69 |
| **MODE** | Transmission Mode for SCGR | Section 7.1.3.2 on page 67 |
| **NUMDEV** | Number of devices | Section 7.1.3.2 on page 67 |
| **OVLTH** | Overload threshold | Section 7.1.5 on page 71 |
| **PACKALG** | Packing algorithm | Section 7.1.7 on page 75 |
| **PAL** | P-GSL timing advance algorithm | Section 7.1.6 on page 74 |
| **PTA** | P-GSL timing advance | Section 7.1.6 on page 74 |
| **PSTU** | Packet switched termination unit | Section 7.1.3.2 on page 67 |
| **SC** | Super channel | Section 7.1.3.2 on page 67 |
| **SCGR** | Super channel group | Section 7.1.3.2 on page 67 |
| **SDAMRREDABISTHR** | Dynamic AMR reduction Abis threshold (AMR FR with reduced codec allocation threshold) | Section 7.1.5 on page 71 |
| **SDFRMAABISTHR** | Dynamic fullrate mode adaption Abis threshold (Dynamic FR to HR mode adaptation threshold) | Section 7.1.5 on page 71 |
| **SDHRAABISTHR** | Dynamic halfrate allocation Abis threshold | Section 7.1.5 on page 71 |
| **SDHRAABISTHRWB** | Dynamic halfrate allocation Abis threshold for AMR-WB capable mobile stations | Section 7.1.5 on page 71 |
| **SDHRMAABISTHR** | Dynamic halfrate mode adaption Abis threshold (Dynamic HR to FR mode adaptation threshold) | Section 7.1.5 on page 71 |
| **SECTOR** | Sector name | Section 7.1.4 on page 69 |
| **SIGDEL** | Signaling delay | Section 7.1.4 on page 69 |
| **STN** | Site transport node for RBS | Section 7.1.3.2 on page 67 |
| **TMODE** | Transmission mode for TG | Section 7.1.3.2 on page 67 |
| **TYPE** | Type of IPADDR | Section 7.1.2 on page 64 |

### 7.1.2 IP Connectivity Parameters

To use IP connectivity, the operator configures the parameters in Table 22. The steps are listed in Reference [19].

Configure an IP address to all PGW-RPs or CTH-RPs with the command RRIPI, and define at least one extra (reconfigurable) IP address per BSC, to support PGW redundancy.

- To configure the IP address of a PGW-RP, give both **IPADDR** and **IPDEVNO** as parameters. The resulting IP address is then fixed to the given PGW-RP (**TYPE** = FIXED).

- To configure the IP address of a CTH-RP, give both **IPADDR** and **APPLSEL** as parameters. The resulting IP address is then reconfigurable by application (**TYPE** = RECAPPL).
- To create a reconfigurable IP address, leave out the parameter **IPDEVNO**. The given **IPADDR** then becomes reconfigurable (**TYPE** = RECONF).
- Define the IP device as a PGW with the parameter **IPDEVTYPE** (use the value in Table 22).
- Define the **IPADDR** sub-net mask with the parameter **MASK**.

Associate Abis over IP to all IP devices with the command RRAPI. This order is valid only for registered applications and remains after restart.

**TYPE** and **IPDEVNO** are printout parameters. **TYPE** is a header that shows whether the **IPADDR** is of type FIXED, RECONF, RECAPPL or RECBW. **DEVNO** is a header for **IPDEVNO** values. **DEVNO** is empty for reconfigurable, reconfigurable best bandwidth and reconfigurable by application IP addresses.

IP Supervision requires parameter configurations in accordance with Reference [22].

*Table 22 IP Connectivity Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **APL** | — | — | String 1–3 | — | “ABI”: Abis over IP |
| **APPLSEL** | — | — | — | — | — |
| **DEVNO**(1) | — | — | 0–511 | Expressed as PGW devices, where n = Device number | — |
| **IPADDR** | — | — | a.b.c.d<br>a: 1-126, 128-223<br>b,c,d: 0-255 | — | — |
| **IPDEVNO** | — | — | 0–511 | Expressed as PGW devices, where n = Device number | — |
| **IPDEVTYPE** | — | — | String 1–7 | — | “RTIPPGW”: for GARP2 based PGW and "RTIPCTH" for EPB1 based PGW. |

*Table 22 IP Connectivity Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **MASK** | — | — | a.b.c.d<br>a: 255<br>b,c,d: 0-255 | — | — |
| **TYPE**(1) | — | — | FIXED, SEMIFIX, RECONF, RECBW, RECAPPL | — | FIXED: The IP address is fixed to the specified RP.<br>RECONF: The IP address may move between RPs at hardware failures.<br>RECBW: The IP address may move between appropriate IP devices at hardware failures. An IP device with the highest bandwidth capability is selected as first choice.<br>RECAPPL: The IP address is moved between appropriate IP devices at hardware failures. An IP devices is selected by application. |

*(1) Read-only parameter*

### 7.1.3 Bundling and Transport Parameters when STN is used

#### 7.1.3.1 Bundling Group Configuration

To set up the bundling group(s), the operator configures the parameters in Table 23. The steps are listed in Reference [19]. Also, some load regulation parameters described in Section 7.1.5 on page 71 are set in the bundling group.

**LBG** is the identity of a bundling group profile. LBG 0 is the default profile, and cannot be removed or changed. An **LBG** profile needs to be created for every different traffic profile desired in BSC. New **LBG** profiles are created with the command RRBGI. Parameter values in existing LBGs (apart from LBG 0) are changed with the command RRBGC.

- Configure the bundling time described in Section 3.3.3 on page 12 with **MCLTDL** (downlink) and **MCLTUL** (uplink). The value specifies the maximum collection time, but is rounded off to the next synchronization interval.

- Configure the maximum IP packet size for the current bundling group with **MPLSDL** (downlink) and **MPLSUL** (uplink). Note that this is the size of the payload part of the IP packets. IP and L2TP headers are not included. This parameters shall normally be set to a as large value as possible without risking fragmentation in the transmission. When setting this value all additional headers that will be added, e.g. IP, L2TP, IPSec, ML-PPP etc., must be considered as well as the MTU of the complete transmission.

- Configure Quality of Service with DiffServ code point values (**DSCPDL** downlink and **DSCPUL** uplink). The default DSCP value is 0, which generates the smallest IP overhead, since all traffic types are bundled together. Each DSCP value defines a separate QoS class and can be used to prioritize packets in the IP network. Configuration of Quality of Service is described in Section 6.6 on page 56 and is not supported by all IP networks.

- Configure whether to calculate the CRC-32 checksum according to ISO 3309 and include it in the IP packet with the parameter **CRC32**. The setting applies to both uplink and downlink transmission. If **CRC32** is ON the IP overhead increases.

**IWDVER** shows the current version number of the BSC-STN IWD used by each TG (see Table 23).

*Table 23 LBG Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **CRC32** | ON | ON | ON/OFF | — | ON: CRC 32 included in IP packet<br>OFF: CRC 32 not included in IP packet |
| **DSCPDL** | 0 | — | 0–63 | — | See Reference [28] |
| **DSCPUL** | 0 | — | 0–63 | — | See Reference [28] |
| **IWDVER**(1) | — | — | 2–6 | — | — |
| **LBG** | 0 | — | 0–255 | — | — |
| **MCLTDL** | 1 | 5 | 1–20 | ms | — |
| **MCLTUL** | 1 | 5 | 1–20 | ms | — |
| **MPLSDL** | 1465 | 1465 | 300–1465 | —- | — |
| **MPLSUL** | 1465 | 1465 | 300–1465 | — | — |

*(1) Read-only parameter*

### 7.1.3.2 Super Channel Group Configuration

To set up the super channels, the operator configures the parameters in Table 24. The steps are listed in Reference [19].

**STN** is the host name the BSC received from each TG Transport connection at BSC-STN link establishment. It is set from OSS via CLI in STN.

Define **PSTU**, a host name within STN that handles the **SCGR**, with the command RRPTI.

- Configure an **LBG** to the **PSTU**.
- Configure **SAPI** values to the **PSTU**.

Define **SCGR**, the group of super channels that terminate in the same TG, with the command RRSGI.

- Set **MODE** = IPM.
- Configure the engineered bandwidth of Abis Upper (**MBWDL** downlink and **MBWUL** uplink) with the help of Reference [10].

- Configure the uplink jitter buffer size **JBSUL** with the help of Reference [29].

Define up to four **SC** numbers per **SCGR** with the command RRSCI. Distribute the TRXs used by one TG evenly over all SCs on the TG.

- The **DCP** defines the port on the BTS that is used by the super channel.
- Configure **NUMDEV**, the number of contiguous pairs of RBLT and RTPGD devices that an SC on Abis Lower is constituted of. BSC allows configuration of an SC which is of another size than this, but then it would be incompatible with the STN hardware. Therefore, the SC size shall always be set to 31 time slots for RBS 2409, and 24, 30 or 31 for SIU or TCU.

Configure other parameters with the command RXMOC:

- Define the **DCP** number used for connection to the LAPD concentrator.
- Change the transmission mode of each TG. Set **TMODE** = SCM.
- Configure the downlink jitter buffer size, **JBSDL** with the help of Reference [29].

*Table 24 SC Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **DCP** | — | — | For BTS port A: Numeral 1 - 31<br>For BTS port B: Numeral 33 - 63<br>For BTS port C: Numeral 287 - 317<br>For BTS port D: Numeral 319 - 349 | This parameter specifies the first termination point in Base Transceiver Station (BTS) for the Super Channel (SC). Several subsequent DCPs are used depending on a value of the parameter NUMDEV | For BTS port A: Numeral 1<br>For BTS port B: Numeral 33<br>For BTS port C: Numeral 287<br>For BTS port D: Numeral 319 |
| **JBSDL** | 20 | 20 | 0–255 | ms | — |
| **JBSUL** | 20 | 20 | 0–255 | ms | — |
| **MBWDL** | — | Use Reference [10] | 64–16384 | kbps | — |
| **MBWUL** | — | Use Reference [10] | 64–16384 | kbps | — |

*Table 24 SC Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **MODE** | — | IPM | IPM/SCM | — | IPM: IP Mode<br>SCM: Super Channel Mode (N/A) |
| **NUMDEV** | — | — | 24, 30, 31 | — | — |
| **PSTU** | — | s1-s2<br>s1: The STN name. Text string 1 - 18 characters.<br>s2: TG Transport Instance ID Numeral 0 - 5. | String 1–20 | — | — |
| **SAPI** | — | — | 0, 10, 11, 12, 62, ALL | — | 0: RSL<br>10: Speech<br>11: CS Data<br>12: (E)GPRS<br>62: OML |
| **SC** | — | 0, 1, 2, 3 | 0–3 | — | — |
| **SCGR** | — | — | 0–2047 | — | — |
| **STN**(1) | — | — | String 1–17 | — | — |
| **TMODE** | — | SCM | TDM/SCM | — | TDM: TG in TDM mode (N/A)<br>SCM: TG in Super Channel mode |

*(1) Read-only parameter*

### 7.1.3.3 LAPD Parameters

Configure LAPD parameters with the command RXMOC:

- Define the **DCP** number used for connection to the LAPD concentrator.

*Table 25 LAPD Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **DCP** | — | — | Numeral 1- 1023<br>A total number of 232 DCPs must be given for CON MO. The given DCPs must be consecutive. | — | — |

## 7.1.4 Bundling and Transport Parameter when Baseband Radio Node is used

To configure the bundling group(s), the operator configures the parameters in Table 26. The steps are listed in Reference [19]..

Bundling is configured in the MO Abis Transport (AT), RXSAT with the command RXMOC.

The AT MO is created automaticaly when the Sector TG MO is created with command RXMOI. All parameters in the AT MO will be initiated by their default values. The **SECTOR** parameter is the name identifying a sector TG in the BSC. This name must be configured with the same value that will be received from the Baseband Radio Node at L2TP link establishment. In the Baseband Radio Node the corresponding parameter is set directly from OSS.

- Configure the bundling time described in Section 3.3.3 on page 12 with **MCLTDL** (downlink) and **MCLTUL** (uplink). The value specifies the maximum collection time, but is rounded off to the next synchronization interval.

- Configure the maximum IP packet size for the current bundling group with **MPLSDL** (downlink) and **MPLSUL** (uplink). Note that this is the size of the payload part of the IP packets. IP and L2TP headers are not included. This parameters shall normally be set to a as large value as possible without risking fragmentation in the transmission. When setting this value all additional headers that will be added, e.g. IP, L2TP, IPSec, etc., must be considered as well as the MTU of the complete transmission.

- For each **SAPI**, configure Quality of Service with DiffServ code point values (**DSCPDL** downlink and **DSCPUL** uplink). The default DSCP setting will result in separate bundling profiles for signalling, (SAPI 0 and 62) with DSCP=51, circuit switched (SAPI 10 and 11) with DSCP=46 and packet switched data (SAPI 12) with DSCP=28. Each DSCP value defines a separate QoS class and can be used to prioritize packets in the IP network. Configuration of Quality of Service is described in Section 6.6 on page 56 and is not supported by all IP networks.

- Configure whether to calculate the CRC-32 checksum according to ISO 3309 and include it in the IP packet with the parameter **CRC32**. The setting applies to both uplink and downlink transmission. If **CRC32** is ON the IP overhead increases.

- Configure the engineered bandwidth of Abis Upper (**MBWDL** downlink and **MBWUL** uplink) with the help of Reference [10].

- Configure the LAPD signalling delay (**SIGDEL** to be used for LAPD layer 2 retransmissions.

Configure other parameters with the command RXMOC:

**IWDVER** shows the current version number of the BSC-STN IWD used by each TG (see Table 23).

*Table 26 Bundling Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **CRC32** | ON | ON | ON/OFF | — | ON: CRC 32 included in IP packet<br>OFF: CRC 32 not included in IP packet |
| **DSCPDL** | 28,46,51 | — | 0–63 | — | See Reference [28] |

*Table 26 Bundling Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **DSCPUL** | 28,46,51 | — | 0–63 | — | See Reference [28] |
| **IWDVER**(1) | — | — | 2–6 | — | Baseband Radio Node will use 6 |
| **MBWDL** | — | Use Reference [10] | 64–16384 | kbps | — |
| **MBWUL** | — | Use Reference [10] | 64–16384 | kbps | — |
| **MCLTDL** | 1 | 5 | 1–20 | ms | — |
| **MCLTUL** | 1 | 5 | 1–20 | ms | — |
| **MPLSDL** | 1465 | 1465 | 300–1465 | —- | — |
| **MPLSUL** | 1465 | 1465 | 300–1465 | — | — |
| **SECTOR** | — | s1-s2<br>s1: The sector name. Text string 1 - 18 characters.<br>s2: Sector id, numeral | String 1–20 | — | — |
| **SIGDEL** | MEDIUM | MEDIUM | NORMAL, MEDIUM, LONG | | "MEDIUM" Parameter value dedicated to Abis over IP overload handling |

*(1) Read-only parameter*

### 7.1.5 Load Regulation Parameters

To make the load regulation described in Section 3.6 on page 15 work effectively, the operator configures the parameters in Table 27 to suit the current transmission network. The engineered bandwidth parameters **MBWDL** and **MBWUL** are already described in Section 7.1.3.2 on page 67 and Section 7.1.4 on page 69.

Overload prevention is described in Section 3.6.3 on page 17 and configured with the command RXATC:

- Set the threshold **SDAMRREDABISTHR** to trigger the feature Full rate AMR on 8 kbps Abis, which restricts AMR codec to max AMR 7.4 kbps over Abis (using AMR FR on the air interface). The recommended value is found in Table 27.

- Activate the feature Fullrate AMR on 8 kbps Abis per cell with **DAMRCRABIS**.

- If the feature Speech Quality Priority is activated, configure **SDAMRREDABISTHR** separately for different subscriber priority levels (see Reference [6]). Use **AHPRL** to define subscriber priority level.

- Configure the feature Abis Triggered HR Allocation with the parameters **SDFRMAABISTHR**, **SDHRAABISTHR**, **SDHRAABISTHRWB**,and

**SDHRMAABISTHR**. They are set in BSC per TG. Recommended values are found in Table 27.

— The threshold **SDFRMAABISTHR** triggers move from full rate to half rate channels of ongoing calls (including AMR) when the bandwidth utilization increases.

— The threshold **SDHRAABISTHR** triggers allocation of half rate channels to new speech calls (including AMR but not AMR-WB) when the bandwidth utilization increases. When the bandwidth utilization decreases below the threshold, half rate allocation stops.

— The threshold **SDHRAABISTHRWB** triggers allocation of half rate channels to new speech calls for AMR-WB capable mobile stations when the bandwidth utilization increases. When the bandwidth utilization decreases below the threshold, half rate allocation stops.

— The threshold **SDHRMAABISTHR** triggers move from half rate (back) to full rate channels of ongoing calls (including AMR) when the bandwidth utilization decreases.

- Activate the feature Abis Triggered HR Allocation per cell with **ATHABIS** (see Reference [6]).
- If the feature Speech Quality Priority is activated, configure **SDHRAABISTHR** separately for different subscriber priority levels (see Reference [6]). Use **AHPRL** to define subscriber priority level.

Overload handling is described in Section 3.6.4 on page 18:

If overload handling is not to be activated set **OVLTH** to 100% and **IPOV** to OFF.

- If overload handling is active configure overload handling with the packet loss threshold **OVLTH** in the LBG profile in BSC with the command RRBGC if STN is used and in the AT MO with command RXMOC is Baseband Radio Node is used.

  — In the most simple case, there is only one **OVLTH** parameter per TG, which is the case for Baseband Radio Node and when Quality of Service is not used in the STN case.

  — In the case an LBG is created for each traffic type in the STN case,, each **OVLTH** parameter will be associated to one traffic type (recommendations for Quality of Service are found in Reference [17]).

  — The packet loss threshold, **OVLTH**, should be configured to a value above the drop rate of the Abis transmission during normal operation (not overload). Note that it is the peak drop rate that should be considered.

  — If the packet drop characteristics of the network are not known **OVLTH** can be tuned in the following manner: set the **OVLTH** parameter to 1000 to turn off this overload trigger. Run the system long enough to obtain drop rate statistics, see chapter Section 5.6.1 on page 32. The statics shows only average drop rate, to obtain the peak drop rate use a 10 time higher value for **OVLTH**. Run the system with the new value of **OVLTH** and continue monitor the drop rate statistics but also monitor the counter OVERLOADREJCON. If OVERLOADREJCON is stepped during low traffic it is most probably due to false overload indication. This means that **OVLTH** is set too low, increase it. Continue the monitoring and repeat until OVERLOADREJCON isn't stepped during low traffic.

**Note:** If the peak drop rate is very low, **OVLTH** should not be set below the value 25 (2.5%).

**Note:** If the peak drop rate is very high it may be necessary to turn the **OVLTH** trigger off by setting it to the value 1000 (100%).

- To switch on LAPD retransmission triggered overload handling, set **IPOV** to ON and **LDEL** to 1 with the command RRSGC in the STN case, or **IPOV** to ON and **SIGDEL** to MEDIUM with the command RXMOC in the Baseband Radio Node case. **LDEL** defines LAPD signalling delay parameters that are equal to **SIGDEL** values NORMAL and MEDIUM.

**Note:** **SIGDEL** have the value NORMAL, which means that this overload handling is not compatible with Abis over satellite for the STN case. No configurable threshold parameters exist.

**IPOV** will start to trigger around 5% packet drop. If it is expected to occasionally have a packet drop rate (i.e. drops not due to overload) above 5% it is recommended to turn **IPOV** off.

*Table 27 Load Regulation Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **ATHABIS** | OFF | ON | ON/OFF | — | — |
| **DAMRCRABIS** | OFF | ON | ON/OFF | — | — |
| **IPOV** | OFF | — | ON/OFF | — | — |
| **LDEL** | 0 | 1 | 0-1 | — | 0: Normal. The LAPD Link parameters are selected by the parameter SIGDEL.<br>1: Medium. Specific LAPD Link parameters are selected. |
| **OVLTH** | 1000 | — | 0–1000 | 1/1000 of all packets in LBG. (25 means 2,5%) | — |
| **SDAMRREDABISTHR** | 100 | 70 | 1–100 | % of **MBWDL** or **MBWUL** | 100: The codec set is restored to default. |
| **SDFRMAABISTHR** | 100 | 85 | 1–100 | % of **MBWDL** or **MBWUL** | 100: No hand-overs are forced. |
| **SDHRAABISTHR** | 100 | 75 | 1–100 | % of **MBWDL** or **MBWUL** | 100: No half-rate allocations are forced. |

*Table 27 Load Regulation Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **SDHRAABISTHRWB** | 100 | 90 | 1–100 | % of **MBWDL** or **MBWUL** | 100: No half-rate allocations are forced. |
| **SDHRMAABISTHR** | 0 | 60 | 0–100 | % of **MBWDL** or **MBWUL** | 0: No returning to full rate is forced. |

### 7.1.6 P-GSL Timing Advance Parameters

**PAL** indicates if the feature Adaptive Timers for Packet Abis is activated. The feature will automatically adjust the timing advance for the Packet-GSL protocol, P-GSL, used by packet Abis.

Note that for Baseband Radio Node the feature Adaptive Timers for Packet Abis is always active.

To activate the feature set **PAL** to 1.

With the feature Adaptive Timers for Packet Abis activated the buffer depth for P-GSL is set by the parameter **JBPTA**. The **JBPTA** value is used by the adjustment algorithm for determining the average usage of a buffer in the TRXs. A setting of **JBPTA** according to the recommendations from Reference [10] and Section 6.3 on page 54 can decrease unnecessary buffer usage and increase end-user performance. A **JBPTA** not set according to the recommendations may decrease (E)GPRS throughput as packets can arrive to the TRXs after their scheduled air time slot has passed. The **PTA** parameter will have no function.

Without Adaptive Timers for Packet Abis (STN case only) the parameter **PTA** indicates a factor for counting the timing advance for P-GSL statically, as in previous releases. The setting of **JBPTA** will then have no function. The **PTA** value is used in BTS to calculate how long time in advance a packet is requested from BSC until it shall be sent over the air interface. It is important to set **PTA** according to the actual round trip time on the BTS - BSC interface. If **PTA** is set too small, packets will arrive to the TRXs after its scheduled air time slot has passed, causing drastically decreased (E)GPRS throughput. If **PTA** is set too large, packets will be placed in a buffer in TRXs and wait for its scheduled air time slot. This buffer can take maximum 6 (RLC/MAC) blocks per time slot. If more than 6 blocks needs queueing, blocks will start to be discarded, causing decreased (E)GPRS throughput. An unnecessary large **PTA** will also cause delays, since blocks are then queued an unnecessary long time. Table 29 below shows what round trip delay each **PTA** value represents, but the correct value to configure is found using the dimensioning guideline: Reference [10].

*Table 28 PTA*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **JBPTA** | 20 | Section 6.3 on page 54 | 0-120 | ms | — |

*Table 28 PTA*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **PAL**<br><br>(not used by Baseband Radio Node) | 0 | 1 | 0,1 | — | 0: The P-GSL timing advance is manually set by the parameter PTA.<br><br>1: Automatic adjustment of the P-GSL timing advance, the parameter JBPTA sets the buffer depth. |
| **PTA**<br><br>(not used by Baseband Radio Node) | 7 | Reference [10] | 0–63 | — | — |

*Table 29 PTA Parameter Values*

| Step Length 1 | | | Step Length 2 | | | Step Length 4 | | | Step Length 8 | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| PTA | Number of frames | Delay (ms) | PTA | Number of frames | Delay (ms) | PTA | Number of frames | Delay (ms) | PTA | Number of frames | Delay (ms) |
| 0 | 1 | 4,6 | 16 | 18 | 83,1 | 32 | 52 | 240,0 | 48 | 120 | 553,8 |
| 1 | 2 | 9,2 | 17 | 20 | 92,3 | 33 | 56 | 258,5 | 49 | 128 | 590,8 |
| 2 | 3 | 13,8 | 18 | 22 | 101,5 | 34 | 60 | 276,9 | 50 | 136 | 627,7 |
| 3 | 4 | 18,5 | 19 | 24 | 110,8 | 35 | 64 | 295,4 | 51 | 144 | 664,6 |
| 4 | 5 | 23,1 | 20 | 26 | 120,0 | 36 | 68 | 313,8 | 52 | 152 | 701,5 |
| 5 | 6 | 27,7 | 21 | 28 | 129,2 | 37 | 72 | 332,3 | 53 | 160 | 738,5 |
| 6 | 7 | 32,3 | 22 | 30 | 138,5 | 38 | 76 | 350,8 | 54 | 168 | 775,4 |
| 7 | 8 | 36,9 | 23 | 32 | 147,7 | 39 | 80 | 369,2 | 55 | 176 | 812,3 |
| 8 | 9 | 41,5 | 24 | 34 | 156,9 | 40 | 84 | 387,7 | 56 | 184 | 849,2 |
| 9 | 10 | 46,2 | 25 | 36 | 166,2 | 41 | 88 | 406,2 | 57 | 192 | 886,2 |
| 10 | 11 | 50,8 | 26 | 38 | 175,4 | 42 | 92 | 424,6 | 58 | 200 | 923,1 |
| 11 | 12 | 55,4 | 27 | 40 | 184,6 | 43 | 96 | 443,1 | 59 | 208 | 960,0 |
| 12 | 13 | 60,0 | 28 | 42 | 193,8 | 44 | 100 | 461,5 | 60 | 216 | 996,9 |
| 13 | 14 | 64,6 | 29 | 44 | 203,1 | 45 | 104 | 480,0 | 61 | 224 | 1033,8 |
| 14 | 15 | 69,2 | 30 | 46 | 212,3 | 46 | 108 | 498,5 | 62 | 232 | 1070,8 |
| 15 | 16 | 73,8 | 31 | 48 | 221,5 | 47 | 112 | 516,9 | 63 | 240 | 1107,7 |

### 7.1.7 Packing Algorithm Parameter

**PACKALG** sets the packing algorithm used on the Abis traffic generated by the TG. The packing algorithm is described in Reference [2]. If Abis over IP is used together with A over IP packing algorithm 1 must be used. For Baseband Radio Node packing algorithm 1 is always used.

*Table 30 Packing Algorithm Parameter*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit | Specific Values |
|---|---|---|---|---|---|
| **PACKALG**<br>(not used by Baseband Radio Node) | 1 | 1 | 0–1 | — | 0: No packing<br>1: packing algorithm 1. |

## 7.2 Main Controlling Parameters in STN

### 7.2.1 Main Controlling Parameters in SIU, TCU and RBS2409

The commands to set parameters and make other configurations in STN are described in Reference [20] for TCU and SIU and in Reference [21] for RBS2409. In the following, the parameters are described per MO class. A class diagram for the MO objects is found in the Managed Object model overview in Reference [23] (for TCU and SIU) and in Reference [24] (for RBS2409).

*Table 31 MO Class E1T1Interface (only one instance supported)*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **type** | R/W | E1_30 | E1 or T1 | E1/ T1/ E1_30/ E1_unframed | - | Type of interface, that is, E1, E1_30, E1_unframed or T1. Value E1_30 defines the use of TS 2–31 (all TSs are used).<br>Rule: A mix is allowed on groups based on 4 Interfaces. The groups are formed based on InstanceId 0-3, 4-7, 8-11, and 12-15.<br>Rule: All 4 links in the group shall either be E1 or T1<br>Rule: E1_30 is only allowed for MO SuperChannel<br>Note: It is allowed to mix E1, E1_30 and E1_unframed in a group. |

*Table 32 MO Class EthernetInterface*

| Parameter Name | CLI | Default Value | Recommended value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **port** | R/W | WANFastEthernet | WANFastEthernet | SFP/ WANGigabit/ WANFastEthernet/ Gigabit/ FastEthernet | - | The port type used for this MO. Rule: SFP can only be used when the istanceId is 0, WAN or CO-LOCATION; WANGigabit and WANFastEthernet can only be used when the instanceId is 0 or WAN; Gigabit can only be used when the instanceId is CO-LOCATION; FastEthernet can only be used when the instanceId is CO-LOCATION or SITE_LAN_n. |
| **mode** | R/W | AUTO | AUTO | AUTO / 10MBithalfduplex/ 10MBitfullduplex/ 100MBithalfduplex/ 100MBitfullduplex/ gigabit | - | Mode in which the interface shall communicate, according to IEEE 802. Rule: gigabit is not a possible if port = WANFastEthernet or FastEthernet. Rule: Only gigabit is possible if port=SFP |

*Table 33 MO Class IPInterface*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **primaryIP_Address** | R/W | notDefined | - | IPv4address | - | The primary IP address of the STN. If IP Security with IPSec protected tunnelling is used, the primary IP address is the outer IP address of the IP packets (it may at the same time also be used for the inner packets as well). Rule: notDefined is not a valid value. When this attribute changes value, all MO instances, which have a depIPInterface attribute that refers to this instance, are affected. Please see the details about the effect under the depIP_Interface attribute of the MOs in question. The relevant MOs are: STN, TGTransport, PingMeasurement, Synchronization, ManualSAPair. |

*Table 33 MO Class IPInterface*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **primarySubNetMask** | R/W | notDefined | - | IPv4subnetMask | - | The subnet mask used by the IP Interface, as set by operator. Rule: notDefined is not a valid value. |
| **MTU** | R/W | 1500 | 1500 | Integer 68..1500 | - | The MTU of the IP interface as set by the operator. |
| **defaultGateway** | R/W | notDefined | - | IPv4address | - | Default gateway used by the IP interface. Rule: Must be defined for the WAN interface (the IPInterface where the depLinkLayer attribute points to an EthernetInterface MO with instanceID “0” or “WAN”). It must have the value notDefined for all other IPInterface objects. |
| **ipDefaultTTL** | R/W | 64 | 64 | Integer 0..255 | - | The default value inserted into the Time_To_Live field of the IP header of datagrams originated at this entity, whenever a TTL value is not supplied by the transport layer protocol. |
| **depLinkLayer** | R/W | STN=0 EthernetInterface=0 | - | MO-DN | - | Reference to the link layer assigned for the IPInterface. Rule: The depLinkLayer can only refer to MO PPPoE or MO EthernetInterface. Rule: Must not refer to the same link layer as the depLinkLayer of a PPPoE instance. |
| **depTrafficManager** | R/W | <empty string> | | MO-DN | | Reference (use dependency) to the traffic manager assigned for the IPInterface. The value <empty string> means that no traffic manager is used for this IPInterface object. Rule: The depTrafficManager can only refer to MO TrafficManager. |

*Table 34 MO Class MeasuredMOClass*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **measuredMO** | N/A | - | - | 1..1024 | - | MO class name of the instance monitored |

*Table 35 MO Class MeasurementCounters*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **counter** | N/A | - | - | String 1..64 | - | The name of the counter attribute to be monitored. |
| **counterPosition** | N/A | - | - | Integer 1..32 | - | Specifies the position in the data file of the measurement report |

*Table 36 MO Class MeasurementDefinition*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **neIndex** | R/W | 0 | - | Integer 0..$2^{16}$ -1 | - | Network element index. |
| **filePrefix** | RO (1) | STN | - | String(m), 0..32 | - | Prefix used in filenames for uploaded files. |
| **granularityPeriod** | RO | - | - | 15 | minutes | The time in minutes between the initiations of two successive gatherings of measurement data. |
| **reportPeriod** | R/W | 15 | 15 | 15 | minutes | Time between the measurement reports are uploaded. |

*(1) The value is defined at creation and afterwards read-only*

*Table 37 MO Class PingMeasurement**

| Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **pingDestination** | R/W | notDefined | - | IPv4address | - | Address of the destination IP host. |
| **pingDSCP** | R/W | 0 | - | Integer 0..63 | - | DSCP code of the ECHO packet. See also Page 67. |
| **depIP_Interface** | R/W | <empty string> | - | MO-DN | - | Reference (usage dependency) to the IP interface to be used for ping measurement. Implicitly the reference also gives the local IP address to use for the ping measurement. When a change to this attribute or the IPAddress attribute of the MO referred to, is committed, this MO will be reconfigured to use the new IP address. Rule: must be one instance of the MO IPInterface or MO SecondaryIP. |
| **pingInterval** | R/W | 107 | - | Integer 0..$2^{16}$-1 | ms | Time between echo requests, measured in ms. pingInterval = 0 means no ping performed. |
| **pingPacketSize** | R/W | 0 | - | Integer 0..$2^{16}$-1 | bytes | Specifies the number of data bytes to be sent. |
| **pingTimeout** | R/W | 20 | - | Integer 4..$2^{16}$-1 | ms | Time-out time in ms for ping |

To enable performance monitoring of the Abis Upper link, the operator creates and configures the PingMeasurement MO. The attribute pingDestination is set to the IP address of the PGW. The attribute pingDSCP is set to the same value as the DSCP of the most sensitive traffic type (normally as CS traffic). The attribute depIPInterface is set to refer to the IP interface used for CS traffic. Note: It is necessary to enable performance monitoring of the Abis Upper Link to get the performance statistics needed for calculation of end-to-end delay and packet loss of the Abis link.

*Table 38 MO Class SecondaryIP*

| Parameter Name | CLI | Default Value | Recommended value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **secondaryIP_Address** | R/W | notDefined | - | IPv4address | - | A secondary IP address of the STN. secondaryIP_Address is configured in order to provide some means to establish a virtual IP address plan. When this attribute changes value, all MO instances, which have a depIPInterface attribute that refers to this instance, are affected. Please see the details about the effect under the depIP_Interface attribute of the MOs in question. The relevant MOs are: STN, TGTransport, PingMeasurement, and Synchronization.<br>If IP Security with IPSec protected tunnelling is used, the secondary IP address is the inner IP address of the IP packets. The secondaryIP_Address must not equal the primaryIP_Address of MO IPInterface.<br>Rule: notDefined is not a valid value |
| **secondarySubNetMask** | R/W | notDefined | - | IPv4subnetMask | - | The subnet mask<br>Rule: notDefined is not a valid value |

*Table 39 MO Class STN*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **depIP_Interface** | R/W | <empty string> | - | MO-DN | - | Reference (usage dependency) to the IP interface assigned for the STN O&M traffic towards the OSS. Implicitly the reference also gives the local IP address to use for communication with the OSS. When this attribute changes value, all MO instances, which have a depIPInterface attribute that refers to this instance, are affected. Please see the details about the effect under the depIP_Interface attribute of the MOs in question. The relevant MOs are: STN, TGTransport, PingMeasurement, and Synchronization.<br>Rule: the MO IPInterface or MO SecondaryIP must be connected to the WAN interface.<br>Rule: must be one instance of the MO IPInterface or MO SecondaryIP. |

*Table 39 MO Class STN*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **wakeUpRegistration** | R/W | notDefined | - | IPv4address/ NOT DEFINED | - | The IP address where STN shall send wake up event. notDefined means that wake up registration is turned off. |
| **wakeUpEventInterval** | R/W | 1 | 1 | Integer 0..10 | minutes | Specifies the value of a timer hold by the STN. It is in unit of whole minute [0..10], where 0 indicates infinity,that is wakeUpMessage is turned off. The STN shall send a wakeUpMessage to the wakeUpRegistration address within this time. Default value is 1, this means that it shall be sent with an interval of 1 minute and stop when a subscribe operation is performed. |
| **DSCP_OperationAndMaintenance** | R/W | 0 | 0 | Integer 0..63 | - | DSCP code for operation and maintenance data between STN and OSS excluding O&M file transfer. See also Section 7.1.3 on page 66. |
| **DSCP_OM_FileTransfer** | R/W | 0 | 0 | Integer 0..63 | - | DSCP code for operation and maintenance file transfer between STN and OSS. File transfer related to PM, BulkCM and SWSession. See also Section 7.1.3 on page 66. |
| **STN_PGW_KeepalivePeriod** | R/W | 1 | 1 | Integer 1..100 | seconds | Period in seconds between keep alive procedures sent from STN to PGW. |
| **STN_PGW_L2TP_Max_Transmissions** | R/W | 5 | 5 | Integer 1..10 | - | Maximum number of retransmissions of L2TP control messages, including the original transmission, according to the L2TP control message retransmission mechanism. This value applies to all control messages, including for example the Setup Control Connection Request message (SCCRQ) and the KeepAlive Hello message. |

*Table 39 MO Class STN*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **STN_PGW_L2TP_RetransmissionCap** | R/W | 4 | 4 | Integer 1..10 | - | Cap on the exponentially increasing interval between retransmission of L2TP control messages, according to the L2TP control message retransmission mechanism. This value applies to all control messages, including for example the Setup Control Connection Request message (SCCRQ) and the KeepAliveHello message. Rule: The STN_PGW_L2TP_RetransmissionCap value must be larger than, or equal to, STN_PGW_L2TP_InitialRetransmissionPeriod. |
| **STN_Name** | R/W | <empty string> | - | String(m) 0..18 | - | Network unique hostname for the STN. Max 18 char long. |
| **STN_PGW_L2TP_InitalRetransmissonPeriod** | R/W | 1 | 1 | Integer 1..10 | - | Initial period from the original transmission to the first retransmission of L2TP control messages according to the L2TP control message retransmission mechanism. This value applies to all control messages, including for example the Setup Control Connection Request message (SCCRQ) and the KeepAlive Hello message. |
| **systemClockTimeServer** | R/W | notDefined | - | IPv4address/ NOT DEFINED | - | The name of the TimeServer which the STN shall use for synchronization of system clock |
| **STN_systemClockUDP_Port** | R/W | 123 | - | Integer 0..65535 | - | The number of the local STN UDP port that the STN shall use for synchronization of system clock |

*Table 40 MO Class SuperChannel*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **depE1T1Interface** | R/W | notDefined | - | Integer $0..2^{16}$-1 | - | Instance number of the MO E1T1Interface there exist a usage dependency to. notDefined is indicated with $2^{16}$-1. Rule: An E1T1Interface instance is only allowed to be used by one and only one instance of SuperChannel. Rule: notDefined is not a valid configuration. |

*Table 41 MO Class Synchronization*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **synchType** | R/W | notDefined | - | timeServer/ notDefined/ clockInput/ /E1T1/none | - | Type of external synchronization source to be used. 'notDefined' means that the external synchronization source is not yet defined, and 'none' means that the STN should not supply synchronization. Rule: if synchType = timeServer there must exist a TimeServer MO to be a valid configuration. Rule: If synchType = E1T1 then there must exist at least one MO E1/T1Interface with receiveClockSource = timingReference. |
| **DSCP_Synchronization** | R/W | 0 | 46 | Integer 0..63 | - | DSCP code for synchronization data between STN and NTP server. See also Section 7.1.3 on page 66. Note: only valid if synchType = timeServer |
| **depIP_Interface** | R/W | <empty string> | - | MO-DN | - | Reference (usage dependency) to the IP interface assigned for synchronization towards the TimeServer. Implicitly the reference also gives the local IP address to use for communication with the TimeServer. When a change to the IPAddress attribute of the MO referred to, is committed, the effect is that the Synchronization MO will be reconfigured to use the new IPAddress. Rule: the MO IPInterface or MO SecondaryIP must be connected to the WAN interface. Rule: must be one instance of the MO IPInterface or MO SecondaryIP. Rule: If the attribute synchType = timeServer this attribute can not be <empty string> |

*Table 42 MO Class TGTransport (the SIU and TCU can support more than one instance)*

| Parameter Name | CLI | Default Value | Recommended Value | Value Range | Unit | Description |
|---|---|---|---|---|---|---|
| **PGW_IP_Address** | R/W | <empty string> | - | IPv4address | - | IP address of the PGW. |

Table 42 MO Class TGTransport (the SIU and TCU can support more than one instance)

| Parameter Name | CLI | Default Value | Recommended Value | Value Range | Unit | Description |
|---|---|---|---|---|---|---|
| **depIP_Interface** | R/W | <empty string> | - | MO-DN | - | Reference (usage dependency) to the IP interface assigned for Abis traffic towards the BSC. Implicitly the reference also gives the local IP address to use for communication with the BSC. When a change to the IPAddress attribute of the MO referred to, is committed, the effect is that prior to the change taking effect, active Traffic channels will be closed down.<br>Rule: the MO IPInterface or MO SecondaryIP must be connected to the WAN interface.<br>Rule: must be one instance of the MO IPInterface or MO SecondaryIP.<br>Rule: shall be the same in all TG transport instances. |
| **overloadReportInterval** | R/W | 1 | 1 | Integer 1..10 | seconds | Overload report interval, in seconds. Time between overload reports sent to BSC if overloadThreshold is exceeded. Value 1 shall be used to make the overload handling in the BSC work. |
| **DSCP_L2TP_CP** | R/W | 0 | - | Integer 0..63 | - | DSCP code for L2TP Control Connection between STN and BSC. See also Section 7.1.3 on page 66. |

Table 43 MO Class TimeServer

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **TS_priority** | R/W | 0 | 0 | Integer 0..$2^{32}$-1 | - | The TimeServer priority. The MO TimeServer with lowest priority value and a working NTP connection shall be used in first hand for synchronization.<br>Hint on usage: make large gaps between priority numbers such that new TimeServer objects may easily be defined and inserted wherever wanted in the priority order.<br>Rule: priority must be unique. |
| **TS_IP_Address** | R/W | notDefined | - | IPv4address | - | The IP address of the TimeServer which the STN shall use for synchronization |
| **STN_TS_UDP_Port** | R/W | 123 | - | Integer 0..65535 | - | The number of the local STN UDP port that the STN shall use for synchronization. |

Table 44 MO Class PPPoE (only for RBS 2409)

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **depLinkLayer** | R/W | STN=0 Ethernetinterface=0 | - | MO-DN | - | Reference to the link layer assigned for PPPoE.<br>Rule: Must not refer to the same link layer as a depLinkLayer of an IPInterface instance. |

*Table 44 MO Class PPPoE (only for RBS 2409)*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **pppAuthenticationProtocol** | R/W | none | - | pap/ chap/ papchap / papchap2/ none | - | Type of authentication protocol allowed towards the PPP access server: pap - Password Authentication Protocol chap - Challenge Handshake Authentication Protocol (MD5 algorithm) papchap - Both PAP and CHAP protocols are allowed, and the authentication credentials are identical for both protocols (pppPassword and pppUsername). papchap2 - Both PAP and CHAP are allowed, but the authentication credentials are different for the two protocols (PAP: pppPassword and pppUsername, CHAP: pppPassword2 and pppUsername2). none - authentication not used. |
| **pppUsername** | R/W | <empty string> | - | String 0..32 | - | Username for PAP and CHAP authentication. Rule: <empty string> is only a valid value if pppAuthenticationProtocol = none is selected |
| **pppUsername2** | R/W | <empty string> | - | String 0..32 | - | User name for CHAP authentication when pppAuthenticationProtocol = papchap2. Rule: <empty string> is a valid value if pppAuthenticationProtocol=none, pap, chap or papchap is selected. |
| **pppPassword** | WO | <empty string> | - | String 0..32 | - | Password for PAP and CHAP authentication. Rule: <empty string> is only a valid value if pppAuthenticationProtocol = none is selected |
| **pppPassword2** | WO | <empty string> | - | String 0..32 | - | Password for CHAP authentication when pppAuthenticationProtocol = papchap2. Rule: <empty string> is a valid value if pppAuthenticationProtocol=none, pap, chap or papchap is selected. |
| **pppIpAddressNegotiation** | R/W | on | - | on/off | - | Enables IP address negotiation with the PPP access server. If negotiation is switched off, the IP address provided by the PPP access server will be ignored. Rule: If IP address negotiation is on, the STN requires that the delegated IP address is equal to the primaryIP_Address of MO Class IPInterface. If not, the STN will terminate the PPP connection and initiate a new connection establishment. |
| **pppMRU** | R/W | 1492 | - | Integer 0/68..1492 | - | The PPP Maximum Receive Unit (MRU) indicates the maximum size in octets that can be handled by the STN. The pppMRU value is requested to be used by the peer side (PPP server). The negotiated size is used by the peer side when sending PPP packets, and in MTU determination for the STN. The value 0 disables MRU negotiation. |
| **pppLinkStatusInterval** | R/W | 10 | - | Integer 0..50 | - | Interval between PPP LCP echo messages (tenth of seconds). 0 means no PPP LCP echo message should be sent. |
| **pppLinkRetries** | R/W | 3 | - | Integer 1..10 | - | Number of reattempts to establish a PPP link connection. |

*Table 44 MO Class PPPoE (only for RBS 2409)*

| Parameter Name | CLI | Default Value | Recommended Value | Type | Unit | Description |
|---|---|---|---|---|---|---|
| **pppLinkRetriesFrequency** | R/W | 5 | - | Integer 5..50 | - | Interval in tenths of seconds for time between pppLinkRetries or time between reattempts to establish a PPP link connection |
| **pppoeACName** | R/W | <empty string> | - | String 0..100 | - | Name of the PPP over Ethernet access concentrator (AC). An empty string indicates that any AC will be accepted. |
| **pppoeSessionReset** | R/W | 0 | - | 0..1440 | - | Daily reset time measured in number of minutes since 00.00 local time. 0 means that the PPPoE session will not be reset. |

### 7.2.2 Main Controlling Parameters in Baseband T605

Abis over IP is using the following Abis over IP specific MO:s and referring to a MO AddressIPv4 instance for the transport on an IP network. The main controlling parameters are described for each of the Abis over IP specific MO:s. For a complete description of the parameters see Reference [23].

*Table 45 MO class TgTransport*

| Parameter Name | Type | Value Range | Default Value | Unit | Description |
|---|---|---|---|---|---|
| bscBrokerIpAddress | IPv4address | - | - | - | The BSC IP address toward which the RadioTNode makes initial contact. The address is specified in dotted-quad decimal notation in accordance with RFC791, without leading zeros. Dependencies: Multicast IP address and broadcast address are not allowed. Disturbances: Changing this attribute causes traffic disturbances. |
| dscpL2tpCp | Integer | 0..63 | 46 | - | DSCP code for L2TP control connection between the TCU and the BSC. Disturbances: Changing this attribute causes traffic disturbances. |
| encapsulation | ManagedObject | - | - | - | Encapsulates the source IP address of the RadioTNode, which must be an AddressIPv4 MO. Used as a source IP address in the L2TP tunnel. Dependencies: All TgTransport MOs must refer to the same AddressIPv4 MO. Disturbances: Changing this attribute causes traffic disturbances. |
| overloadReportInterval | Integer | 1..10 | 1 | seconds | Number of seconds between overload reports sent to the BSC, if packet loss exceeds overloadThreshold value. |
| overloadThreshold | Integer | 0..10000 | - | - | Threshold for reporting packet loss, measured in hundredths of one percent. The attribute reflects the configuration made by the BSC. |
| stnPgwKeepAlivePeriod | Integer | 1..100 | - | seconds | Number of seconds between keepalive messages sent from the RadioTNode to the BSC. |
| stnPgwL2TpInitalRetransPeriod | Integer | 1..10 | 1 | seconds | Initial period from the original transmission to the first retransmission of L2TP control messages (in seconds), in accordance with the L2TP control message retransmission mechanism. This value applies to all control messages, including the SCCRQ and the keepalive Hello message, for example. Dependencies: The value must be less than or equal to the value of stnPgwL2TpRetransCap. |

*Table 45 MO class TgTransport*

| | | | | | |
|---|---|---|---|---|---|
| stnPgwL2TpMaxTransmissions | Integer | 1..10 | 5 | - | Maximum number of retransmissions of L2TP control messages, including the original transmission, in accordance with the L2TP control message retransmission mechanism. This value applies to all control messages, including the Setup Control Connection Request message (SCCRQ) and the keepalive Hello message, for example. |
| stnPgwL2TpRetransCap | Integer | 1..10 | 4 | seconds | Specifies the cap on the exponentially increasing interval between retransmission of L2TP control messages (in seconds), in accordance with the L2TP control message retransmission mechanism. This value applies to all control messages, including the SCCRQ and the keepalive Hello message, for example. Dependencies: The value must be greater than or equal to the value of stnPgwL2TpInitialRetransPeriod. |
| tgTransportId | String | - | - | - | The value component of the RDN. For a working configuration, the value of tgTransportId must be identical to the network-unique name for the RadioTNode as stated in the BSC (attribute PSTU) which is restricted to 20 characters. The recommended naming of PSTU in the BSC is a concatenated string with the syntax "BSC:STN name-BSC:PSTU number[0..5]". Example: "NewYorkCentral-1". |

*Table 46 MO Class SuperChannel*

| Parameter Name | Type | Value Range | Default Value | Unit | Description |
|---|---|---|---|---|---|
| encapsulation | ManagedObject | - | - | - | Reference to the underlying encapsulation, which can be an E1T1Port MO. Dependencies: A maximum of one SuperChannel MO can refer to the same E1T1Port MO. |
| superChannelId | String | - | - | - | The value component of the RDN. Dependencies: The value of supperChannelId must be 0, 1, 2 or 3. |

*Table 47 MO Class E1T1Port*

| Parameter Name | Type | Value Range | Default Value | Unit | Description |
|---|---|---|---|---|---|
| admOperatingMode | enum | E1 | E1 | - | Specifies the administrative operating mode of the interface. |

*Table 47 MO Class E1T1Port*

| | | | | | |
|---|---|---|---|---|---|
| encapsulation | ManagedObject | - | - | - | Reference to the underlying encapsulation, which is an EtPort MO. Dependencies: A maximum of two E1T1Port MOs can refer to the same EtPort MO. portInterfaceNumber and encapsulation must be unique. Encapsulation must refer to EtPort. |
| portInterfaceNumber | Integer | 0..1 | 0 | - | The EtPort interface number to which this E1T1Port MO is mapped. Dependencies: The attributes encapsulation and portInterfaceNumber together specify the E1T1Port MO. portInterfaceNumber and encapsulation must be unique. |

*Table 48 MO Class EtPort*

| Parameter Name | Type | Value Range | Default Value | Unit | Description |
|---|---|---|---|---|---|
| etPortId | String | - | - | - | The value component of the RDN. The value must match the corresponding connector label on the unit (any spaces are replaced by underscore). Dependencies: The value must be ET_A, ET_B, ET_C or ET_D. |

*Table 49 MO Class TwampTestSession*

| Parameter Name | Type | Value Range | Default Value | Unit | Description |
|---|---|---|---|---|---|
| dscp | Integer | 0..63 | 0 | - | DSCP value of TWAMP IP test packet. |
| dstIpAddress | string | - | - | - | Destination IP address of the TWAMP responder. |
| dstUdpPort | Integer | - | - | - | Destination UDP port of the TWAMP responder. |
| mode | enum | LIGHT LIGHT_SL | LIGHT | - | Type of responder mode of the measurement done in the towards direction. |
| operationalState | enum | DISABLED ENABLED | - | - | The operational state. |
| payload | Integer | 41..1472 | 50 | byte | TWAMP test packet payload length. Example: If ethernet interface IPv4 traffic is tagged and the payload is set to 50, the IP packet size =78 and ethernet frame size =100. If ethernet interface IPv6 traffic is tagged and the payload is set to 50, the IP packet size =98 and ethernet frame size =120. |
| profileType | enum | FULL_RESOLUTION_50 FULL_RESOLUTION_10 | FULL_RESOLUTION_10 | - | TWAMP profile type. |

*Table 49 MO Class TwampTestSession*

| | | | | | |
|---|---|---|---|---|---|
| srcIpAddress | Managed Object | - | § | - | Source IP address. Valid reference is an AddressIpv4 MO. |
| srcUdpPort | Integer | 4032..4063 | - | - | Source of the UDP port to send and receive TWAMP test packets. |

## 7.3 Main Controling Parameters in Baseband Radio Node

Abis over IP is using the following Abis over IP specific MO:s and referring to a MO AddressIPv4 instance for the transport on an IP network. The main controlling parameters are described for each of the Abis over IP specific MO:s. For a complete description of the parameters see Reference [23].

*Table 50 MO class AbisIp*

| Parameter Name | Type | Value Range | Default Value | Unit | Description |
|---|---|---|---|---|---|
| bscBrokerIpAddress | IPv4address | - | - | - | The BSC IP address toward which the Radio Node makes initial contact. The address is specified in dotted-quad decimal notation in accordance with RFC791, without leading zeros. Dependencies: Multicast IP address and broadcast address are not allowed. Disturbances: Changing this attribute causes traffic disturbances. |
| dscpSectorControlUL | Integer | 0..63 | 46 | - | DSCP code for L2TP control connection between the Radio Node and the BSC. Disturbances: Changing this attribute causes traffic disturbances. |
| ipv4Address | Managed Object | - | - | - | Reference to the MO AddressIPv4 which represents the IPv4 address which the GSM Sector uses. |
| keepAlivePeriod | Integer | 1..100 | · | seconds | Number of seconds between keepalive messages sent from the Radio Node to the BSC. |
| initalRetransmissionPeriod | Integer | 1..10 | 1 | seconds | Initial period from the original transmission to the first retransmission of L2TP control messages (in seconds), in accordance with the L2TP control message retransmission mechanism. This value applies to all control messages, including the SCCRQ and the keepalive Hello message, for example. Dependencies: The value must be less than or equal to the value of retransmissionCap. |

Table 50 MO class AbisIp

| | | | | | |
|---|---|---|---|---|---|
| maxRetransmissions | Integer | 1..10 | 5 | - | Maximum number of retransmissions of L2TP control messages, including the original transmission, in accordance with the L2TP control message retransmission mechanism. This value applies to all control messages, including the Setup Control Connection Request message (SCCRQ) and the keepalive Hello message, for example. |
| retransmissionCap | Integer | 1..10 | 4 | seconds | Specifies the cap on the exponentially increasing interval between retransmission of L2TP control messages (in seconds), in accordance with the L2TP control message retransmission mechanism. This value applies to all control messages, including the SCCRQ and the keepalive Hello message, for example. Dependencies: The value must be greater than or equal to the value of stnPgwL2TpInitialRetransPeriod. |
| gsmSectorName | String | - | - | - | The gsmSectorName is used as identification when BSC is connected. The gsmSectorName is sent as attribute HostName in L2TP message SCCRQ. The same name must be configured in the BSC. Default gsmSectorName is the same as gsmSectorId, last 20 characters. |

Table 51 MO Class GsmSector

| Parameter Name | Type | Value Range | Default Value | Unit | Description |
|---|---|---|---|---|---|
| gsmSectorId | String | - | - | - | The value component of the RDN. Sent to the BSC for Sector TG to GSM Sector correlation purposes. |

Table 52 MO Class TwampTestSession

| Parameter Name | Type | Value Range | Default Value | Unit | Description |
|---|---|---|---|---|---|
| dscp | Integer | 0..63 | 0 | - | DSCP value of TWAMP IP test packet. |
| dstIpAddress | string | - | - | - | Destination IP address of the TWAMP responder. |
| dstUdpPort | Integer | - | - | - | Destination UDP port of the TWAMP responder. |
| mode | enum | LIGHT<br>LIGHT_SL | LIGHT | - | Type of responder mode of the measurement done in the towards direction. |
| operationalState | enum | DISABLED<br>ENABLED | - | - | The operational state. |

*Table 52 MO Class TwampTestSession*

| | | | | | |
|---|---|---|---|---|---|
| payload | Integer | 41..1472 | 50 | byte | TWAMP test packet payload length. Example: If ethernet interface IPv4 traffic is tagged and the payload is set to 50, the IP packet size =78 and ethernet frame size =100. If ethernet interface IPv6 traffic is tagged and the payload is set to 50, the IP packet size =98 and ethernet frame size =120. |
| profileType | enum | FULL_RESOLUTION_50 FULL_RESOLUTION_10 | FULL_RESOLUTION_10 | - | TWAMP profile type. |
| srcIpAddress | Managed Object | - | § | - | Source IP address. Valid reference is an AddressIpv4 MO. |
| srcUdpPort | Integer | 4032..4063 | - | - | Source of the UDP port to send and receive TWAMP test packets. |

## 7.4 Main Controlling Parameters in BTS

**CRC-4** is an installation parameter in the BTS Installation Data Base (IDB), used to activate or deactivate Cyclic Redundancy Check 4 (CRC 4) handling. **CRC-4** is set by OMT at installation, and is normally not changed after initial configuration. The consequence of an incorrect setting may be that the whole transmission network goes down.

The CRC 4 procedure provides enhanced error-monitoring capability. CRC 4 is only applicable for type E1 transmission interfaces. **CRC-4** is configured to match the CRC 4 setting in the transport network and most operators use the same CRC 4 setting for their entire network. **CRC-4** is set for every new RBS with E1 transmission.

The **STN Equipment** parameter specifies if there is communication between the RBS and STN or not. In Abis over IP the value must be set to SIU.

*Table 53 BTS Parameters*

| Parameter Name | Default Value | Recommended Value | Value Range | Unit |
|---|---|---|---|---|
| **STN Equipment** | - | SIU | SIU, No | — |
| **CRC-4** | Deactivated | Activated | Activated, Deactivated | — |

# Glossary

The following glossary includes both abbreviations, acronyms, and single word terms:

**Abis**
*node interface*

The Abis interface is the interface between BSC and BTS.

**ADSL**
Asymmetrical Digital Subscriber Line

**AGW**
A-Interface GateWay

AGW handles IP transmission of User Plane Data on the A-interface

**AMR**
Adaptive Multi Rate

**Baseband Radio Node**
A multi standard capable node including BTS, NodeB and/or eNodeB functions realized by hardware for support system, Multistandard Baseband Unit and Radio Units.

**BIP**
BTS Internal Protocol

**BRAS**
Broadband Remote Access Server

**BSC**
Base Station Controller

**BSC IP network interface**
Used as a common name for the BSC LAN Switch and the BSC NWI-E.

**BSS**
Base Station Subsystem

**BTS**
Base Transceiver Station

**Bundling**
*IP optimization technique*

Bundling refers to the packing of several frames into the same IP packet. Frames that share the same end points and bundling parameters forms a bundling group. It is possible to bundle for example one single traffic type (SAPI) into one bundling group. It is possible to get Quality of Service by configuring different priority levels (DiffServ) for different traffic types. When using an STN, the bundling is configured by using bundling profiles (LBGs) and requires that the operator creates one LBG for each priority level. When using a Baseband Radio Node, the bundling is configured per traffic type.

DiffServ implements the IP functionality for differentiated services, or Quality of Service, which allows the IP packets to be distinguished by any receiving router or node. DiffServ is a value of an optional field in the IPv4 and IPv6 header. Packets with higher DiffServ value are forwarded ahead of packets with lower DiffServ value.

**Bundling Group**
An L2TP transport session between the BSC and STN or Baseband Radio Node, used for Abis traffic transportation. All traffic types (SAPIs) in a transport session are bundled into the same IP packet and share the same end points and bundling parameters. When using an STN, there is one bundling group per LBG. When using a Baseband Radio Node, the number of bundling groups depends on configuration and the Baseband Radio Node's TRX allocation in hardware.

**CF**
Central Function

**CLI**
Command Line Interface

**CP**
Central Processor

**CRC**
Cyclic Redundancy Check

**CS**
Circuit Switched

Speech is CS, but also data can be circuit switched, and is then called CS data.

**CTH**
Combined Traffic Handler

RP application in Evo Controller 8200 hosting the TRH, AGW and PGW functions

**DiffServ**
*See* Bundling

**DL**
Down Link

DL is the direction from BSC to BTS.

**DSCP**
Differentiated Services Code Point

DSCP is equal to DiffServ. *See* Bundling

**DSL**
Digital Subscriber Line

ADSL is a type of DSL (*see* ADSL).

**dTRU**
Double Transceiver Unit

A board with two TRUs

**DTX**
Discontinuous Transmission

**E1**
*transmission format*

European digital transmission format 1 for PCM connections [2.048 Mbits/s, 32 independent 64kbps channels, time slots 0-31]

**EFR**
Enhanced Full Rate

**ESP**
Encapsulating Security Payload

**FER**
Frame Erasure Rate

**FR**
Full Rate

**FF**
Frame Format

FF is the highest protocol layer in Abis over IP.

**GEM**
Generic Ericsson Magazine

**GESB**
Gigabit Ethernet Switch Board

**GPH**
GPRS Packet Handler.

GPH is the RP parts of the PCU.

**GPRS**
General Packet Radio Service

**GPS**
Global Positioning System

**GS**
Group Switch

**GSL**
GPRS signaling Link

**GSM**
Global System for Mobile communications

**HDLC**
High level Data Link Controller

HDLC is a protocol that operates at the data link layer (layer 2) of the Open System Interconnection (OSI) architecture. All frames start and end with the flag ='01111110'. The frames also contain a check sum. HDLC contains many other protocols, like LAPD, LAPB and SDLC.

**HR**
Half Rate

**HW**
Hardware

**IDB**
Installation Data Base

**IETF**
Internet Engineering Task Force

**IP**
Internet Protocol

**IPM**
IP Mode

**IPSec**
IP Security

**IPv4**
Internet Protocol version 4

**IRP**
Integration Reference Point

IRP (Integration Reference Point): an architectural concept that is described by a set of specifications for definition of a certain aspect of the Itf-N, comprising a Requirements specification, an IRP Information Service specification, and one or more IRP Solution Set specifications.

**L2**
Layer 2

**L2TP**
Layer 2 Tunneling Protocol

**LAN**
Local Area Network

**LAN switch**
*network interface*

All transmission on Abis Upper goes through the LAN switches to and from BSC. The LAN switches are the network interface of BSC.

**LAPD**
Link Access Procedure on the D-channel

LAPD is a layer 2 protocol used for signaling on the Abis interface. The initial purpose with LAPD was to convey information between layer 3 entities across the ISDN user-network interface using the D-channel. Since then, LAPD has been used for many types of purposes, one of them is Abis Signaling. LAPD provides a transparent connection between two layer 2 peers. The frames contain address and data information, and other things. For further details see ITU-T recommendation Q.921 (I.441).

**LBG**
LAPD Bundling Group

*See* Bundling

**LNS**
L2TP Network Server

**MAC**
Medium Access Control

**Macro RBS**
*See* SIU

**MIB**
Management Information Base

MIB is the logical structure of an SNMP interface.

**ML-PPP**
Mult Link Point to Point protocol

**MO**
Managed Object

**MS**
Mobile Station

Cellular telephone

**MTU**
Max Transmission Unit

**NEDSS**
Network Element Distributed Support Server

**NESS**
Network Element Support Server

**NI**
Network Interface

NI refers to the BSC LAN Switches or the BSC NWI-E.

**NTP**
Network Time Protocol

**NWI-E**
Network Interface - Ethernet

An board-based implementation of the BSC IP network interfaces.

**O&M**
Operations & Maintenance

**OMT**
Operations & Maintenance Terminal

**OSS**
Operation and Support System

**Packet Abis**
*collection name*

Packet Abis is the collection name for the features Packet Abis over TDM and Packet Abis over IP.

**Payload**
*type of traffic*

Payload is the transport load that users may pay for, which is speech and data.

**Pico RBS**
RBS 2409

RBS 2409 is a single TG Pico RBS with an integrated STN for termination of IP and with an high-stable oscillator for timing reference.

**ping**
*IP network command*

The ping command is normally realized with the ICMP echo request/response message. The ping command contacts a unit in an IP network. The IP address is given as parameter. The response shows whether the unit was reached or not.

**P-GSL**
Packet GPRS Signaling Link

P-GSL is a packetizing GSL. P-GSL Timing Advance *see* PTA.

**PCM**
Pulse Code Modulation

**PCU**
Packet Control Unit

**PGW**
Packet GateWay

PGW handles the termination of Packet Abis in the BSC

**PPB**
Parts Per Billion

**PPM**
Parts Per Million

**PPP**
Point to Point Protocol

**PPPoE**
PPP over Ethernet

**PS**
Packet Switched

**PSTU**
Packet Switched Termination Unit

**PTA**
P-GSL Timing Advance

PTA is the time the CCU need to send a DL Data Request in advance in order to cope with the variable Abis delay.

**QoS**
Quality of Service

**RAN SEGW**
*See also* SEGW

A SEGW (firewall) located at the BSC site to provide a secure IPSec tunnel towards the RBS site through an external, unsecure transport network. This functionality is provided by an external equipment.

**RBS**
Radio Base Station

**RBS SEGW**
*See also* SEGW

A SEGW (firewall) located at the RBS site to provide a secure IPSec tunnel towards the BSC site through an external, unsecure transport network. This functionality is provided by the STN node.

**R/W**
Read / Write

**RO**
Read Only

**Round trip time**
*See* RTT

**RP**
Regional Processor

**RTT**
Round Trip Time

RTT is the time it takes to send something back and forth between two nodes. It is measured by sending a request and wait for the response. A one way delay can often be assumed be the RTT divided by 2.

**SAPI**
Service Access Point Identifier

**SC**
Super Channel

**SCM**
Super Channel Mode

**SEGW**
Security Gate Way

SEGWs are network nodes optimized for security handling. They manage VPN tunnels and offer extensive firewall capabilities. Read *User Description, GSM RAN IP Security* for further information regarding SEGWs and BSS IP security issues.

**SFTP**
Secure File Transfer Protocol

**SIU**
Site Integration Unit

SIU is a hardware used to implement the STN for Macro base stations.

**SLA**
Service Level Agreement

This is the agreement with the internet service provider. SLA bandwidth refers to the bandwidth agreed.

**SMO**
Software Management Organizer

SMO is an application in OSS.

**SMRS**
Software Management distribution Repository Services

The OSS software repository.

**SNMP**
Simple Network Management Protocol

**SSH**
Secure SHell

**STN**
Site Transport Node for RBS

STN is the logical name for network equipment used by RBSs for IP RAN transport.

**SW**
Software

**T1**
*transmission format*

T1 is the american standard for 1.544Mbps PCM connections carrying 24 independent 64 kbps channels (DS0) numbered 1-24.

**TCU**
Transport Connectivity Unit

TCU is a hardware used to implement the STN for Macro base stations.

**TDM**
Time Division Multiplexing

TDM refers to the classic synchronous transmission of GSM traffic with a time slot dedicated to each connection.

**TEI**
Terminal Endpoint Identifier

TEI is the address in the IP packet header.

**TG**
Transceiver Group

A TG is the object in BSC that maps a BTS.

**TimeServer**
*node in Abis over IP*

The TimeServer is a network element used to provide synchronization signaling. The TimeServer is fed by a reference clock signal. The clients are provided time stamped signaling with for example NTP.

**TLS**
Transport Layer Security

**TRA**
Transcoder Rate Adapter

The TRA is a unit in BSC that handles speech and CS data.

**TRH**
Transceiver Handler

The TRH is a unit in BSC that handles signaling.

**TRX**
Transceiver

The TRX corresponds to one TRU in the BTS. In the BSC object model a TRX corresponds to one TRXC, one RX, one TX, and eight TSs.

**TWAMP**
Two Way Active Measurement Protocol

**UDP**
User Datagram Protocol

**UL**
Up Link

UL is the direction from BTS to BSC.

**URI**
Uniform Resource Identifier

**VAF**
Voice Activity Factor

VAF is the percentage speech frames compared to total number of speech and silence frames one way in a connection.

**VPN**
Virtual Private Network

**WAN**
Wide Area Network

A WAN is, unlike LAN, a distributed network, with nodes located in separate geographical places.

**XML**
Extensible Mark up Language

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