Nokia Siemens Networks

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RG20 (BSS) Network Engineering Information
A over IP
BSS21341

Piotr Malatynski
Network Engineering
NWS LTE RA E2E Mgmt SA NE GSM & LTE Migration
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A over IP NEI

Content

A over IP NEI

Dependency Table

GSM 800	GSM 900	GSM 1800	GSM 1900	MSS	MGW	NetAct	BSC	SGSN
Y	Y	Y	Y	M15 (System Release 3.3)	Y (System Release 3.3)	OSS5 5.2 CD set 3	S15**	-

Ultra Site	Metro Site	Talk Family	Flexi EDGE	Flexi Multiradio	BTSplus
Y*	Y*	N	Y*	Y*	Y*

*for the scenario with transcoding located in Core Network (MGW) either Flexi EDGE or Flexi Multiradio are required

** BSC3i 660 is not supported

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AoIP overview

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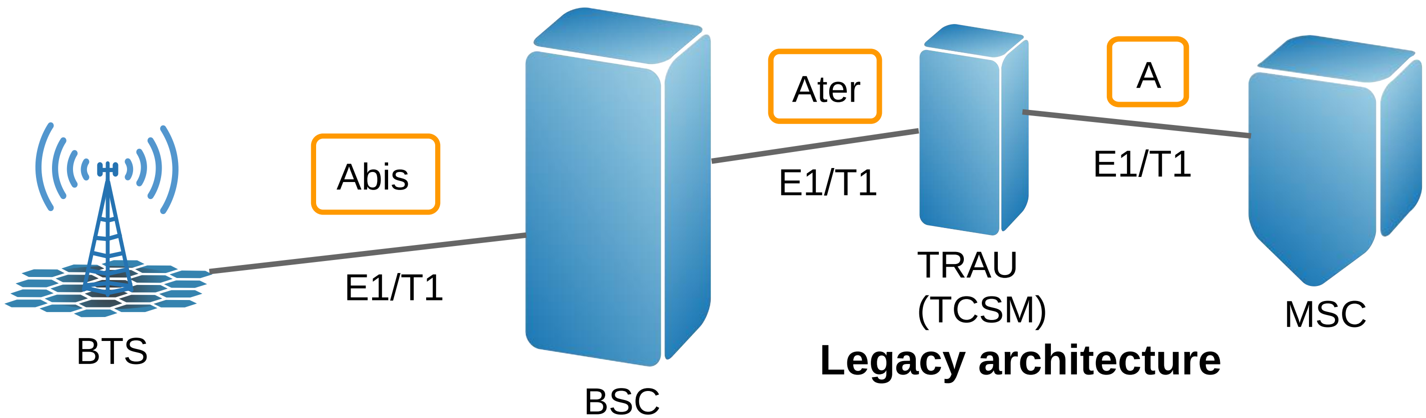
Introduction

NSN CS access network evolution 1/3

- Base Station Subsystem (BSS) consists of BTS, BSC and TRAU
- To connect BSS with CS Core Network A interface is used
- A interface (and all other BSS interfaces) is realized in legacy architecture over **E1 or T1 TDM links**
- Signaling traffic is realized through **SS7 (Signaling System no. 7) messages over E1/T1 TDM links**

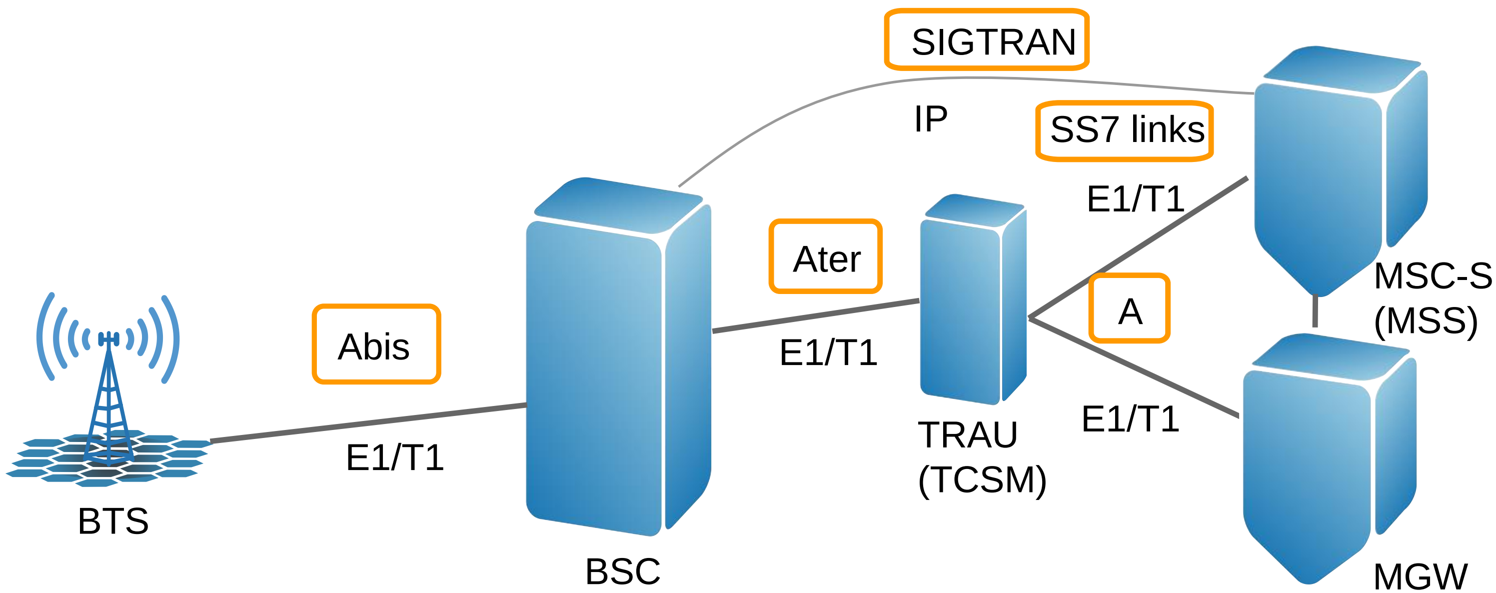


Legacy architecture

Introduction

NSN CS access network evolution 2/3

- Since 3GPP release 4 CS core is split into **Media Gateway (MGW) and MSC Server (MSS)**
- New elements separate call control and user traffic into two independent streams
- MSC Server handles call control and MGW user traffic
- 3GPP rel. 4 introduces also SS7 signaling over IP protocol stack (SIGTRAN)

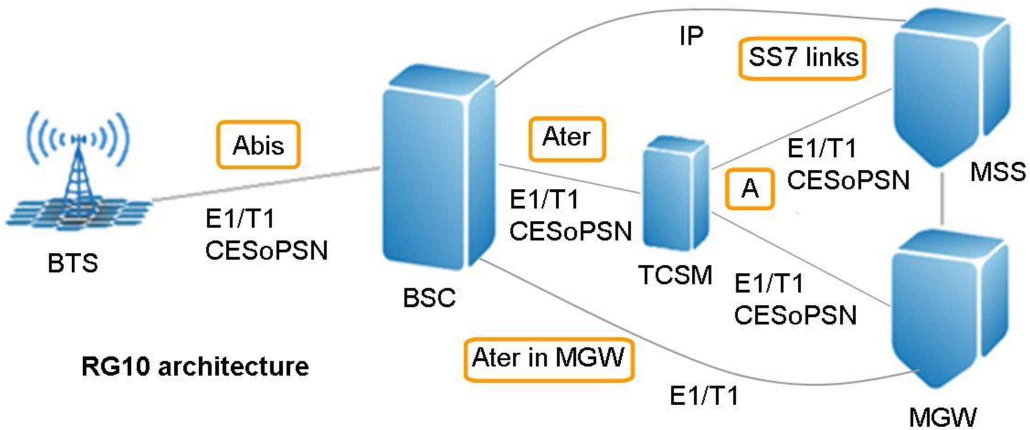


3GPP release 4 architecture

Introduction

NSN CS access network evolution 3/3

- MGW and MSS are available since M12/U2 Core release
- Since S11.5 there is proprietary Ater in MGW solution available that allows to connect TDM based Ater directly to NSN MGW
- SIGTRAN is available in NSN solution since S12 ED
- Since BSS13 Abis, Ater and A interfaces can be realized also by CESoPSN
- CESoPSN is integrated solution since RG10



- RG20 introduces native IP connectivity on Abis interface (Packet Abis feature)
- **RG20 introduces possibility of usage native IP connectivity also in user plane of A interface (AoIP feature)**

Introduction

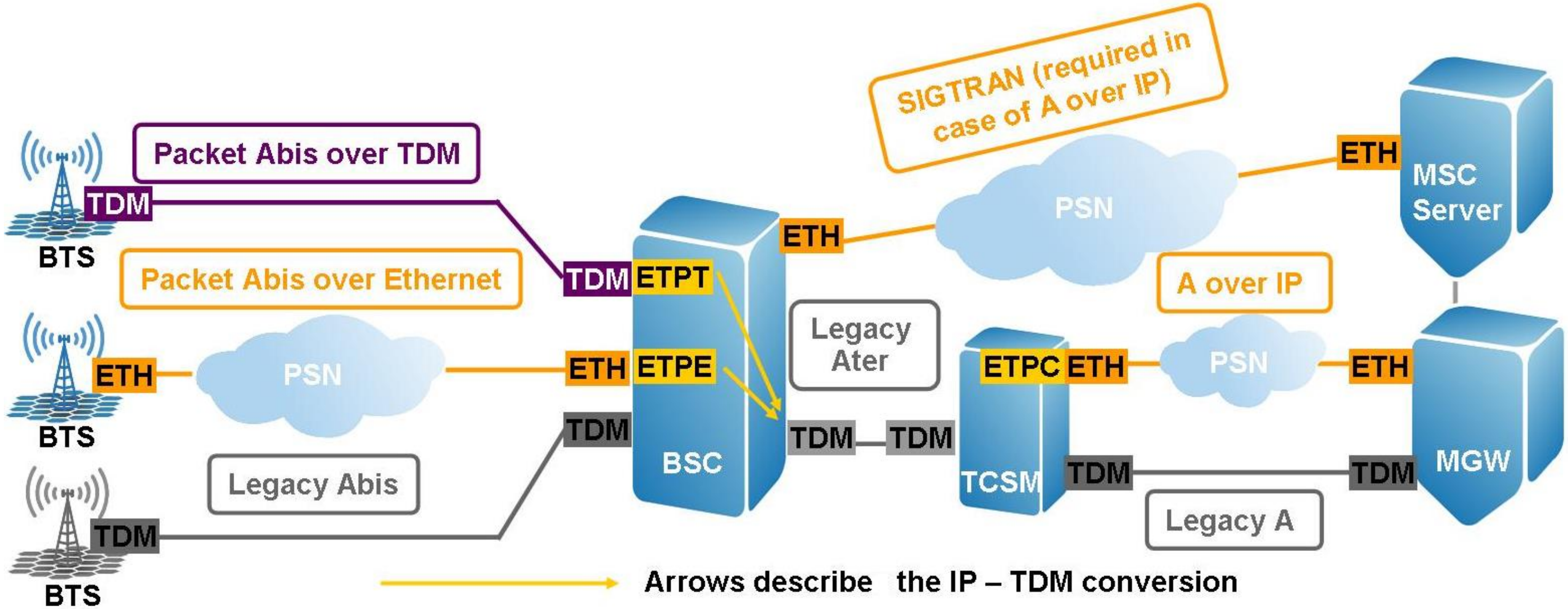
AoIP overview

- In **AoIP** feature frames on A interface between BSC and MGW are transmitted in PSN:
  - **Each frame is a subject to packetization** process prior to sending it to the transmission path
  - as a result of packetization the **incoming frames** are **encapsulated into IP packet payloads** and transmitted **every 20ms** over Ethernet based A interface
  - There is no multiplexing in AoIP – each frame is a single packet
- AoIP is standardized in 3GPP (any vendor CN eligible to supports IP based A interface can work with NSN AoIP solution)
- Due to removal of PCM lines between BSC and CN, synchronization chain is broken and new methods are required for generation and distribution of synchronization signal in GERAN, for details incl. methodology and relevant features please refer to Packet Abis Synchronization, A. Maciolek
- AoIP feature can be realized in **RG20** in two ways:
  - AoIP with Transcoding in BSS
  - AoIP with Transcoding in MGW

Introduction

AoIP possible implementation (basic description) 1/2

- AoIP with Transcoding located in BSS:
  - In U – plane IP connectivity is established between TCSM and MGW
  - In C-plane SIGTRAN (SS7 over IP) feature is used between BSC and MSC server
  - Ater is in use as in previous releases
  - Any realization of Abis interface is possible
  - Legacy A and AoIP implementations can coexist in the same TCSM



Ref. A. Vainio, AoIP SFS ver 1.3.0

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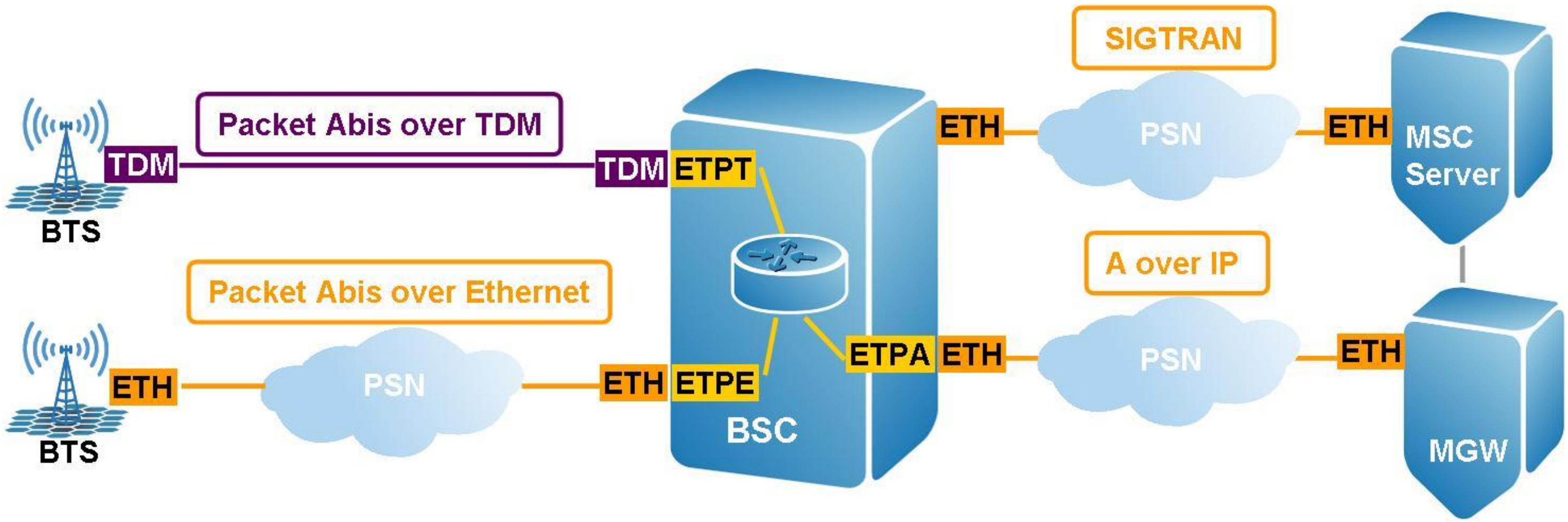
Introduction

AoIP possible implementation (basic description) 2/2

- AoIP with Transcoding located in MGW:
  - In U – plane IP connectivity is established between BSC and MGW
  - In C-plane SIGTRAN (SS7 over IP) feature is used between BSC and MSC server
  - There is no Ater – transcoding functions are moved to MGW
  - Packet Abis (over Ethernet or over TDM) is required in this configuration



Ref. A. Vainio, AoIP System Feature Specification, ver 1.3.0

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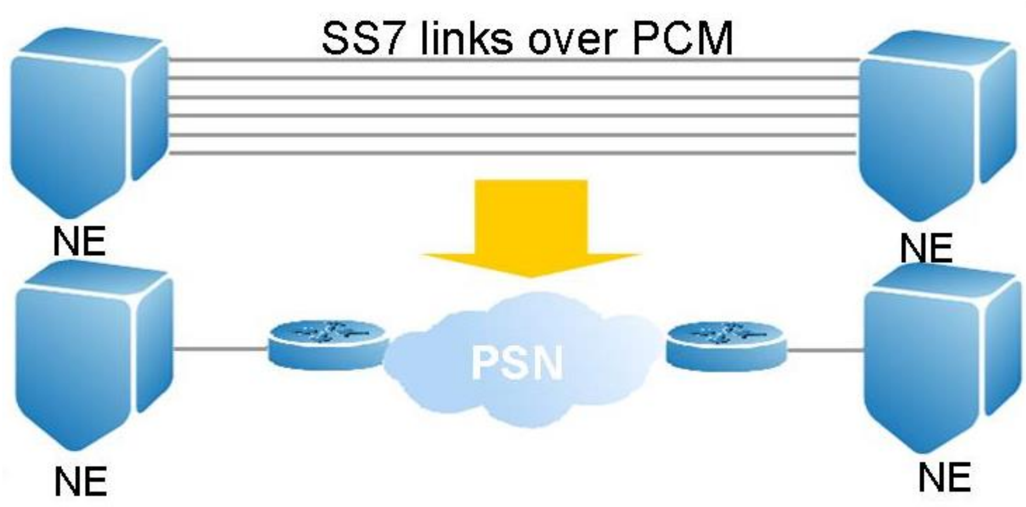
Introduction

SIGTRAN overview

- In AoIP feature there are no PCM lines between CN and BSS
- That means that only possible SS7 signalization realization is SIGTRAN which has some important advantages related to legacy SS7:
  - IP based transport is cost efficient
  - Signaling link capacity is not tied to a fixed bandwidth
  - PSN based transport is more resistant to equipment failures than E1/T1 lines
- Moreover introduction of SIGTRAN on A interface enables usage of the same backbone for all kind of traffic
- Because BSS20852 SIGTRAN feature was introduced in S12ED release for more information, planning and dimensioning rules please refer to existing documentation



Introduction

SW requirements

- Software requirements:
  - The feature (and license key for) BSS21323, L3 Connectivity for Flexi BSC product family is required
  - In case of AoIP with TC in CN(MGW), BSS21231 Packet Abis is required
  - **In U-plane each AoIP scenario needs its own license:**
    - License for AoIP configuration, where TC is located in BSS
    - License for AoIP configuration, where TC is located in CN
  - Each license must be active on the both sides of the connection

Introduction

HW requirements

- Hardware requirements:
  - **FlexiBSC Product Family (excluding BSC3i 660) and TCSM3i** (if TC is in BSS) are needed for AoIP feature
  - New Hardware Unit is required for terminating AoIP U-plane both in BSC3i and TCSM3i and that unit is called **ETP-A** – Exchange Terminal (ET) for Packet transport over A interface
  - Two new functional unit that handles ETP-A are called:
    - **ETPC** (located in TCSM and used in AoIP with TC in BSS case)
    - **ETPA** (located in BSC and used in AoIP with TC in MGW case)
  - ETPC uses 2n redundancy model (two ETPC makes a pair with one in active status)
  - ETPA uses n+1 redundancy model (each group of ETPA has one redundant ETPA unit)
  - BSC C-plane (SIGTRAN) is connected in FlexiBSC product family into **LAN Switching Unit** (SWU) equipment (existing functionality)
  - With the ETPA, the LAN switch extension for BSC3i 1000/2000 and Flexi BSC with the latest LAN switch version (ESB24-D) is necessary
- For more information about HW requirements please refer to Packet Abis NEI, M. Grygiel.

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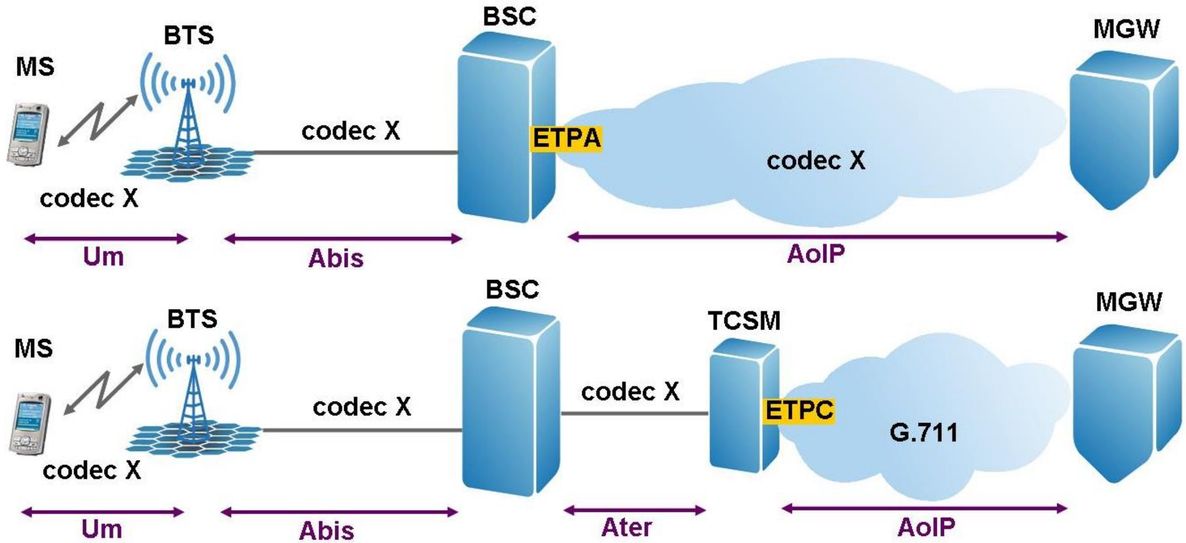
AoIP realization

Realization

Introduction 1/2

- AoIP realization option (TC in BSS or in MGW) determines:
  - Abis interface realization
    - TC in MGW requires Packet Abis to be used
    - TC in BSS can be used with any of Abis interface realizations (although Packet Abis is not recommended – see next slide)
  - Required bandwidth
    - Radio codecs* in case of TC in MGW transmitted across A interface have much **lower bandwidth** requirements than G.711 codec used in Legacy and AoIP with TC in BSS realizations

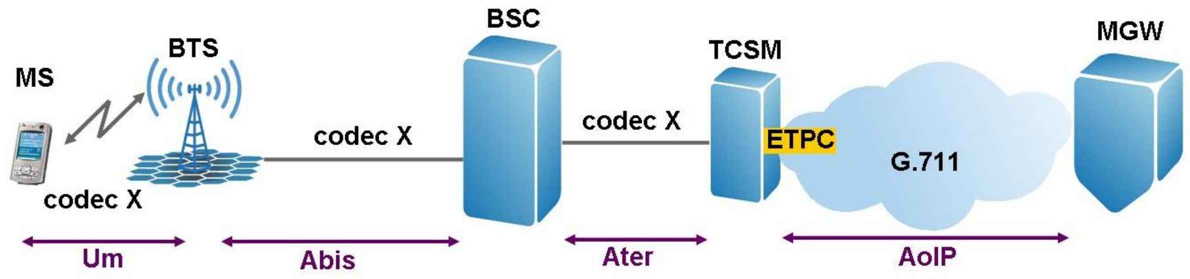


*radio codecs = bits from source encoder e.g. 4.75kbps

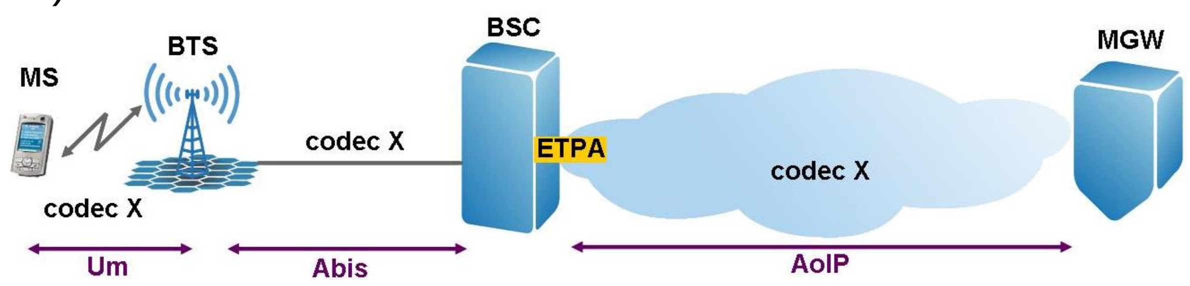
Realization

Introduction 2/2

– Delay
  - Each conversion from PCM to IP (or from IP to PCM) introduces additional delay (because of that Packet Abis together with AoIP with TC in BSS is not recommended)
  - De-jittering in ETPC (TC in BSS) introduces additional delay



  - Removal of TCSM from network architecture (TC in MGW) **improves e2e delay**
  - Transcoder free Operation in case of TC in MGW **improves e2e delay** (see next slides)



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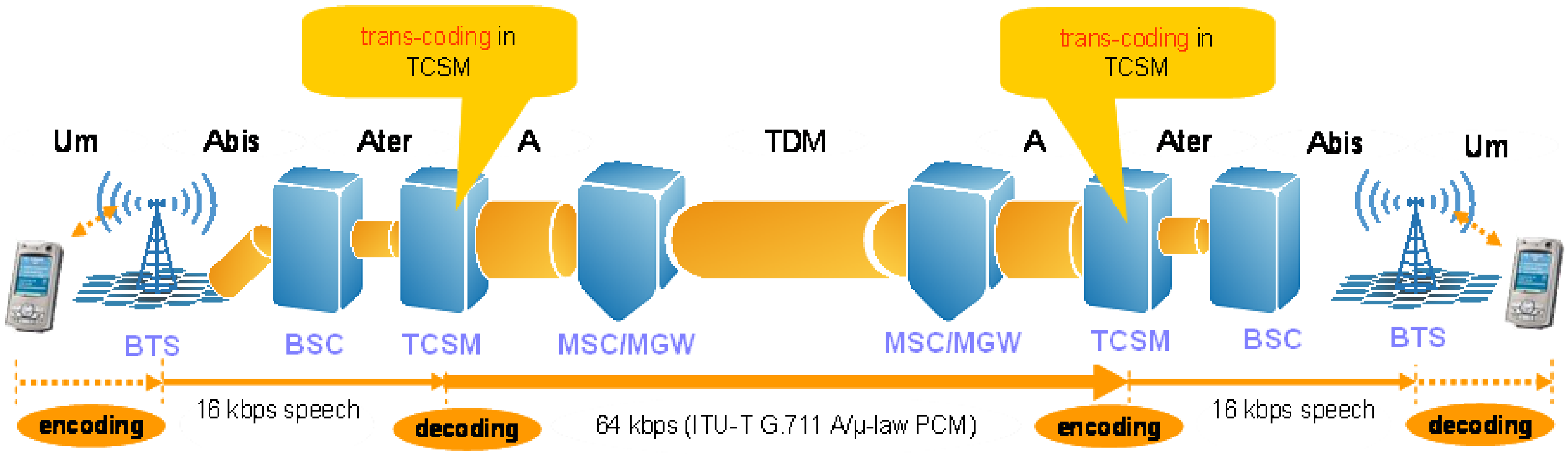
Realization

TFO and TrFO 1/4

- **In Legacy A realization the only possible codec is PCM (G.711)**
- To convert speech codecs (16 or 8kbps) from radio interface to the 64kbps G.711 transcoding is needed and such operation is called tandem operation
- **Tandem operation** degrades speech quality by doubling encoding/decoding



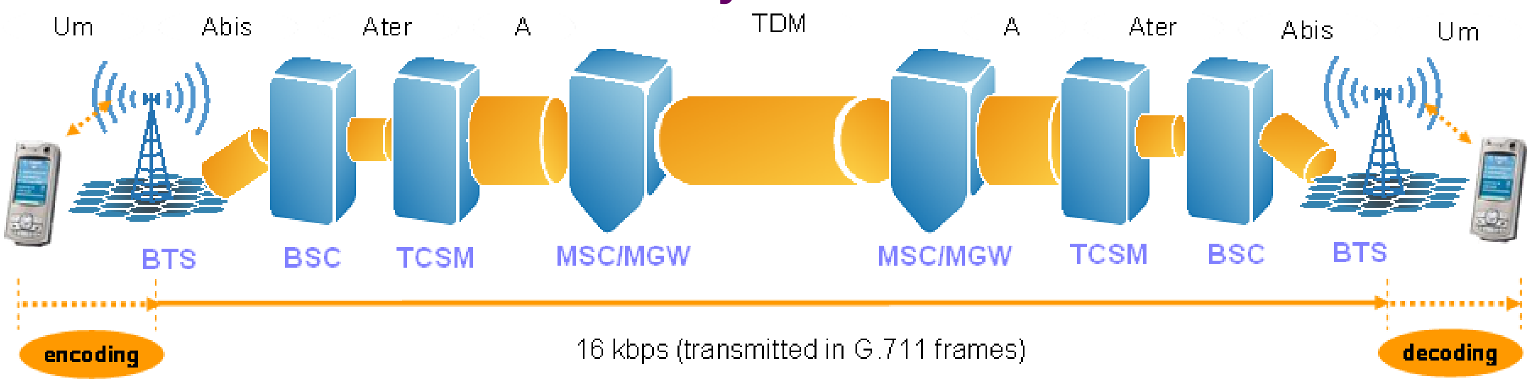
Ref. A. Maciolek, Wideband AMR & AMR TFO for BSS

- To avoid that degradation the Tandem Free operation is possible (TFO) when both mobile stations use the same codec type
- TFO support for all codec types is possible since RG10

Realization

TFO and TrFO 2/4

- TFO aims at avoiding the double speech trans-coding in the MS-to-MS calls
- If both MSs use the same codec type it is convenient to send frames with that codec transparently through the network – then the speech quality degradation due to transcoding is avoided
- TFO is established after call setup and in-band signalling is used (stealing bits)
- Speech codecs bits are transmitted on 2 (16kbps) or 1 (8kbps) least significant bits of G.711 codec frame whereas the remaining G.711 bits are encoded according to A/μ-law
- **In case of AoIP TFO is used only when TC is done in BSS**



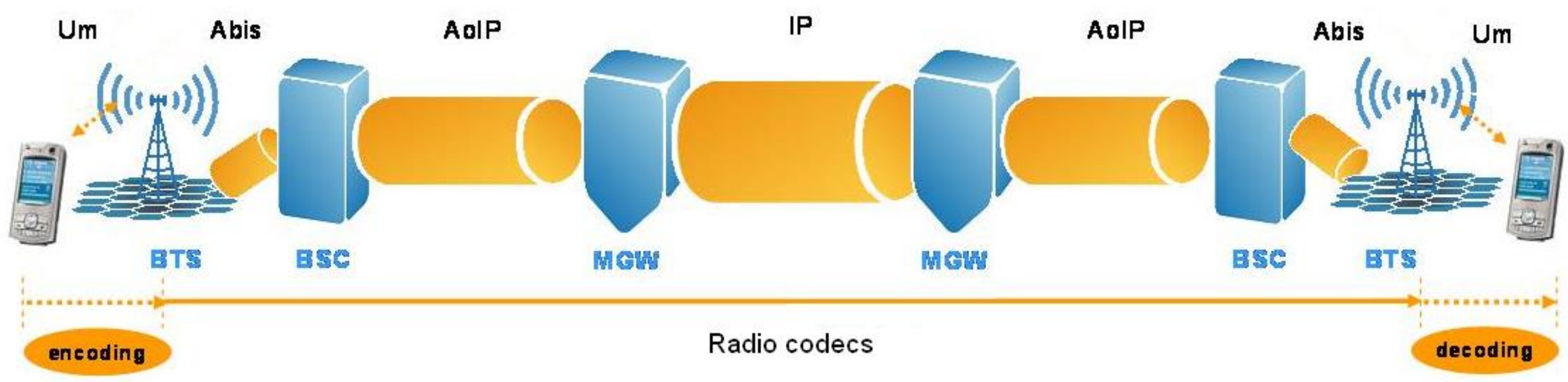
Ref. A. Maciolek, Wideband AMR & AMR TFO for BSS

- For more information about TFO please refer to AMR Wideband & TFO NEI, slide 31

Realization

TFO and TrFO 3/4

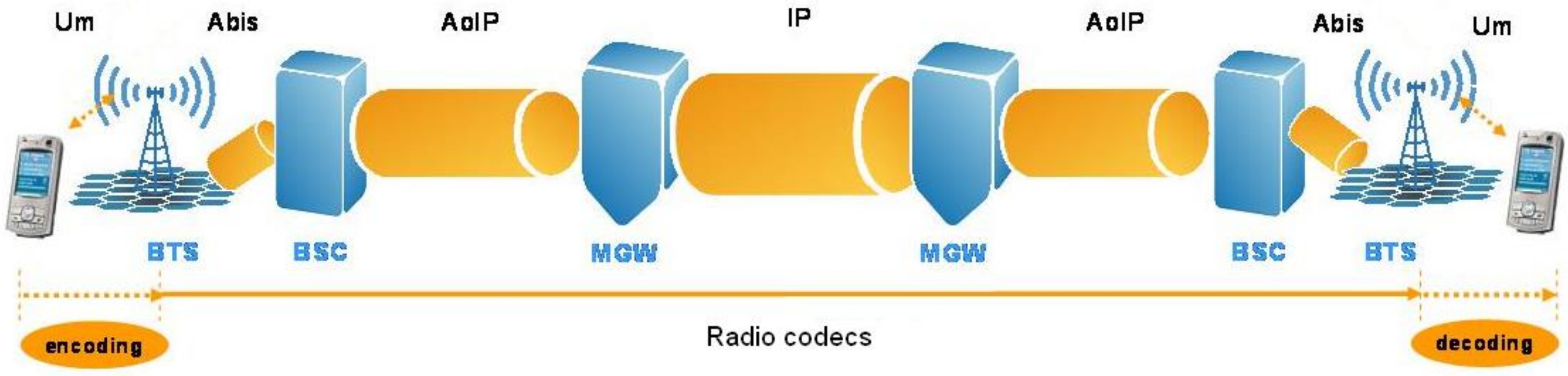
- In case of AoIP with transcoding in MGW it is possible to use Transcoder free Operation (TrFO) instead of TFO
- TrFO also aims at avoiding double transcoding in the transmission path
- There are following differences between TrFO and TFO:
  - MSS decides during call setup procedure which codec configuration (codec type and interface realization) will be used in call
  - Because of that extra delay in call setup is expected (measurements of call setup procedure with TrFO is needed)
  - TrFO utilizes out of band signaling (SIGTRAN) between BSC and MSS
  - G.711 are not present on A interface



Realization

TFO and TrFO 4/4

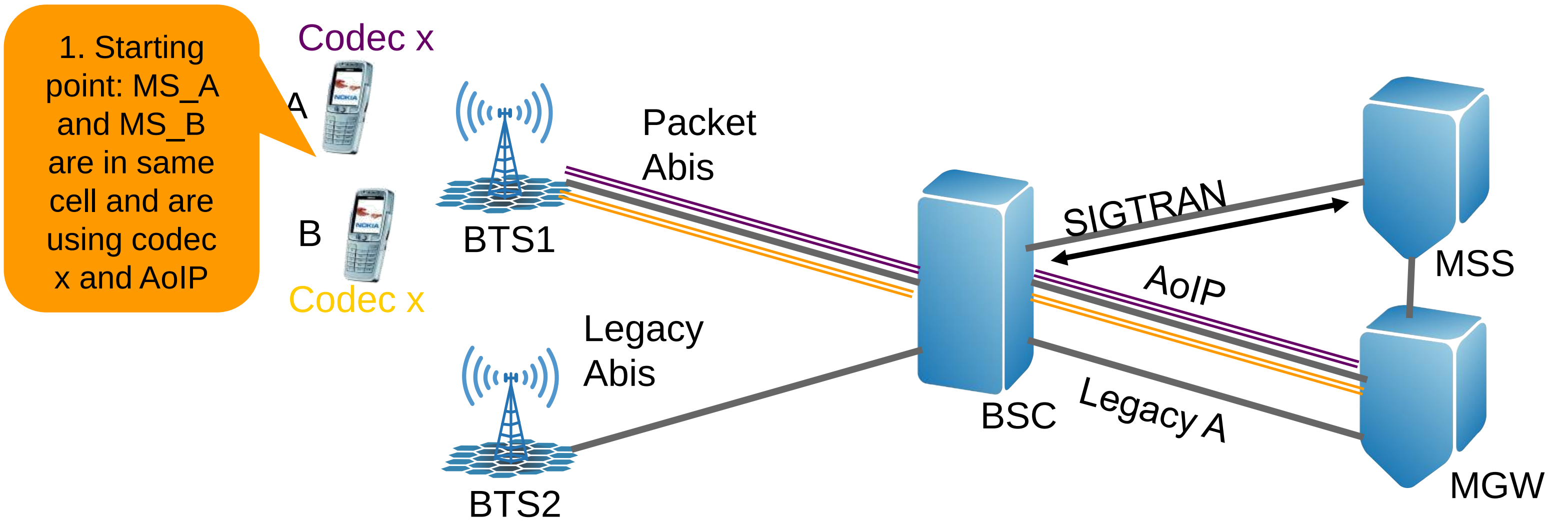
- If TrFO is not possible (e.g. different supported codec types at both connection sides) one of MGW on call path is responsible for transcoding but the second side of the connection is not aware about that
- TrFO or TFO **are mandatory** in case of MS-MS AMR-WB calls
- Both TrFO and TFO improves speech quality, but in addition TrFO reduces delay and improves bandwidth efficiency



Realization

MSS controlled Intra-BSC Handovers 1/2

- In cases when A interface type, codec type and/or codec source bitrate must to be changed during Intra-BSC handover there are needed MSS controlled Intra - BSC Handovers
- They are needed to maintain TFO or TrFO on A interface after handover
- They require new message flows on SIGTRAN which increases signaling load (more information can be found in dimensioning and planning aspects section)
- Case 1: Mismatch of supported A interfaces:



Realization

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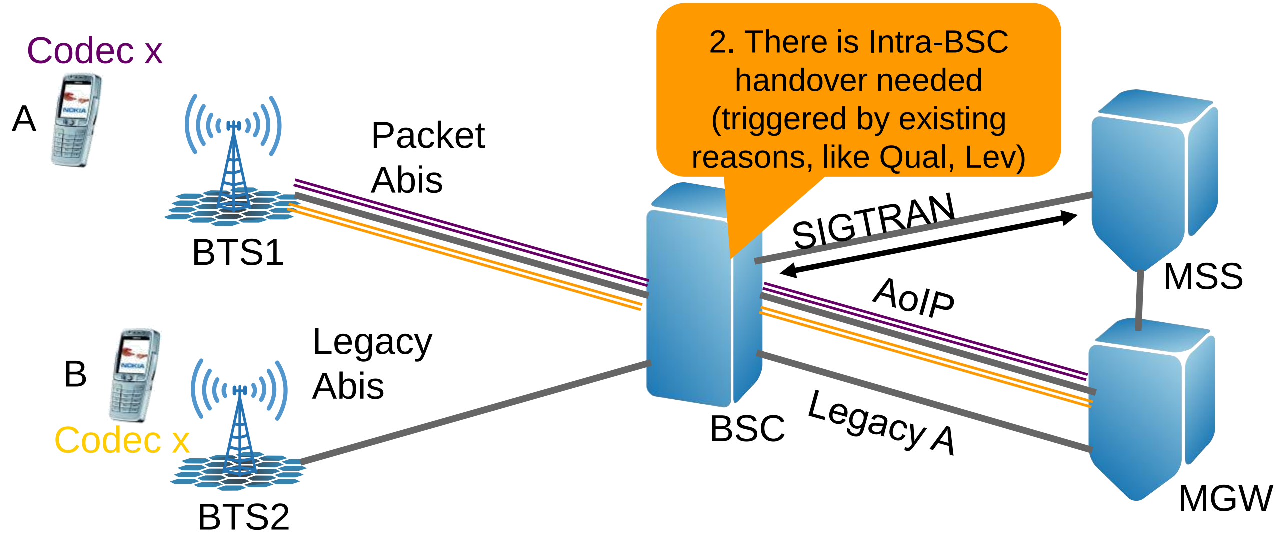

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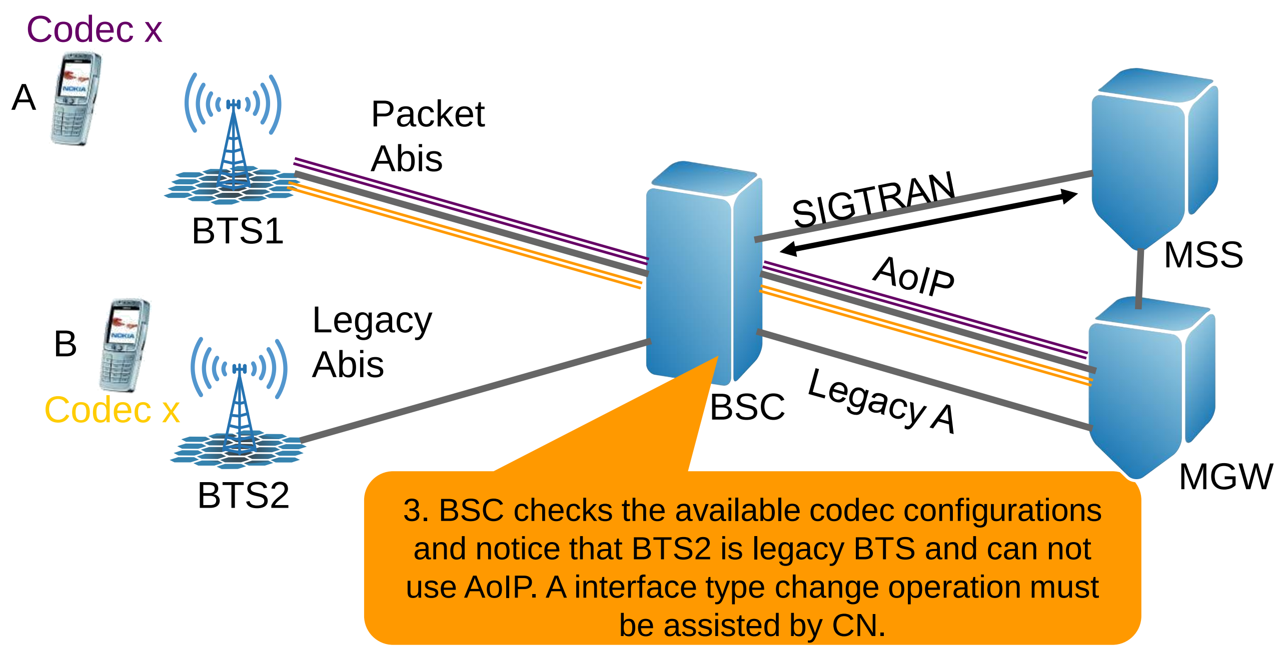

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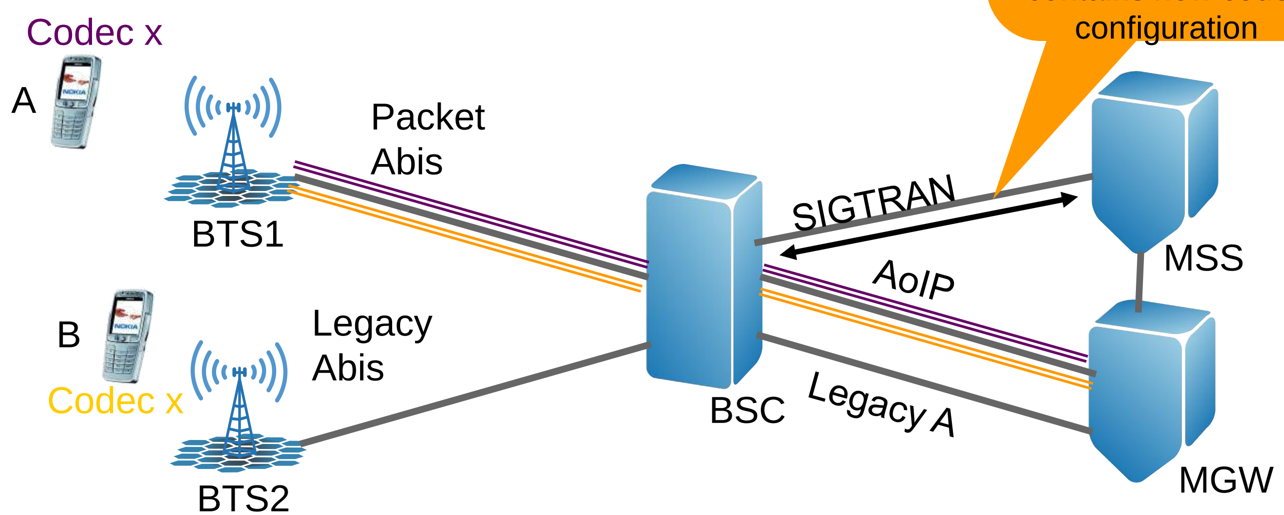

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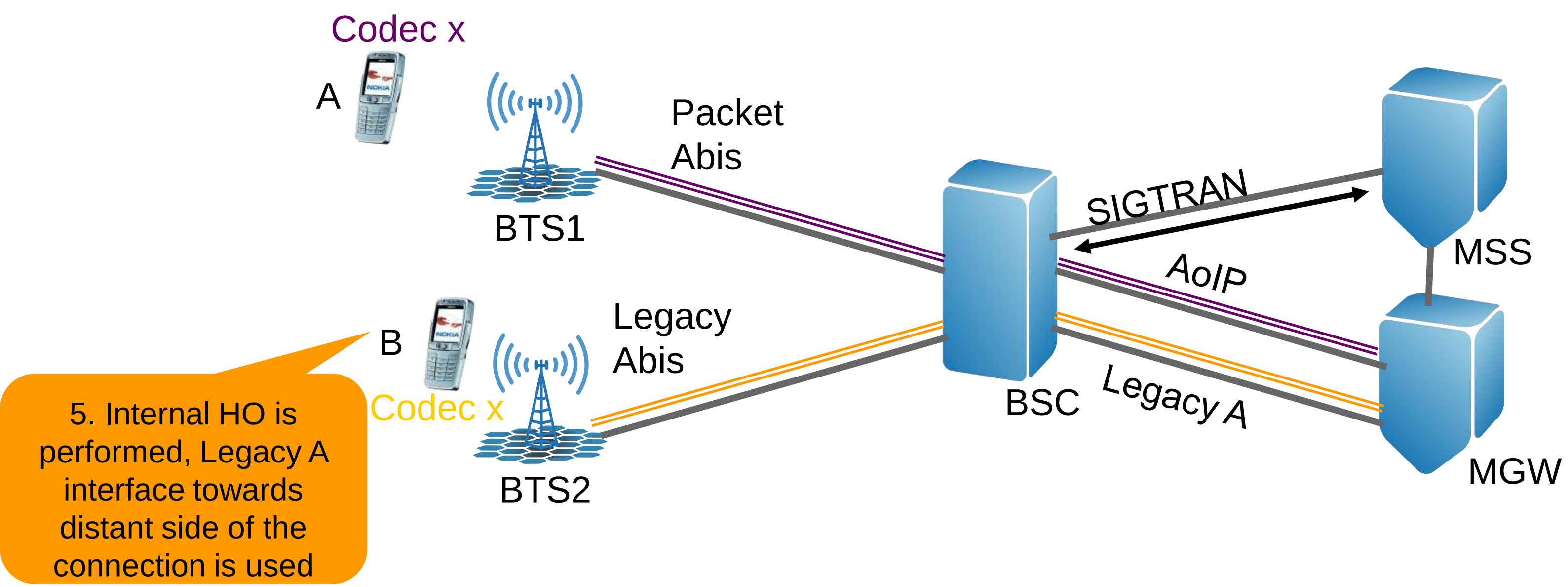

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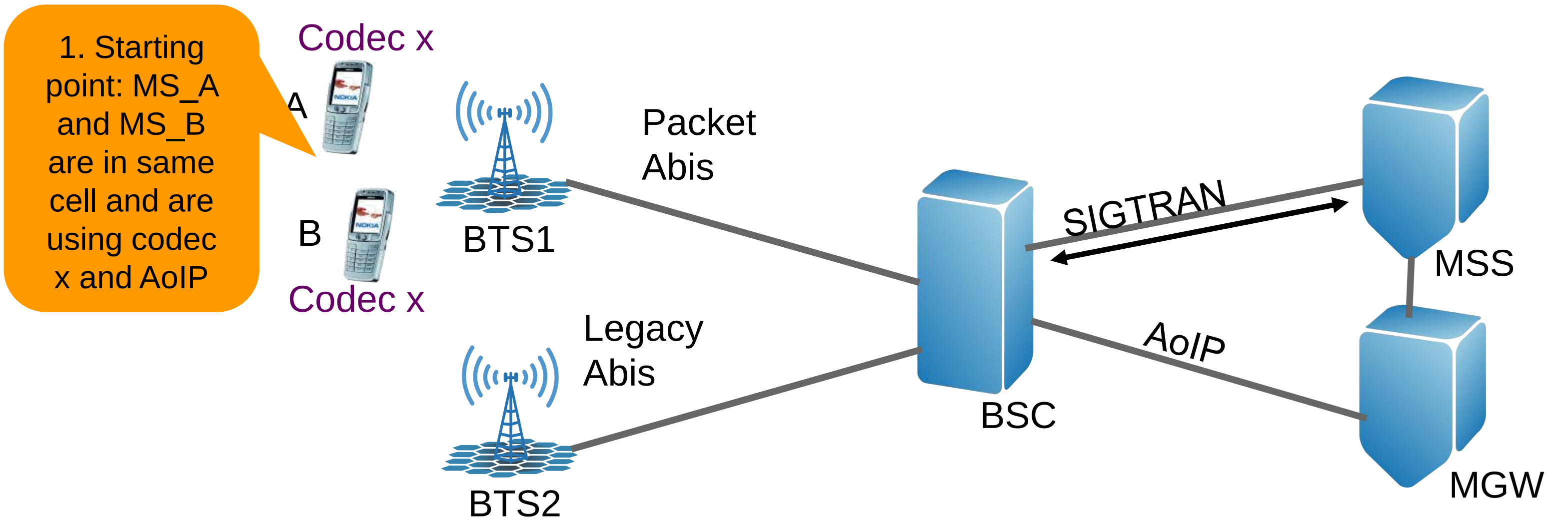
Realization

MSS controlled Intra-BSC Handovers 2/2

- Case 2: codec type or codec source bitrate change
- In case of codec type or codec source bitrate change in distant side of connection there is also needed Internal HO (forced by MSS) to compatible configuration in local side to assure that TFO or TrFO is still possible



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Realization

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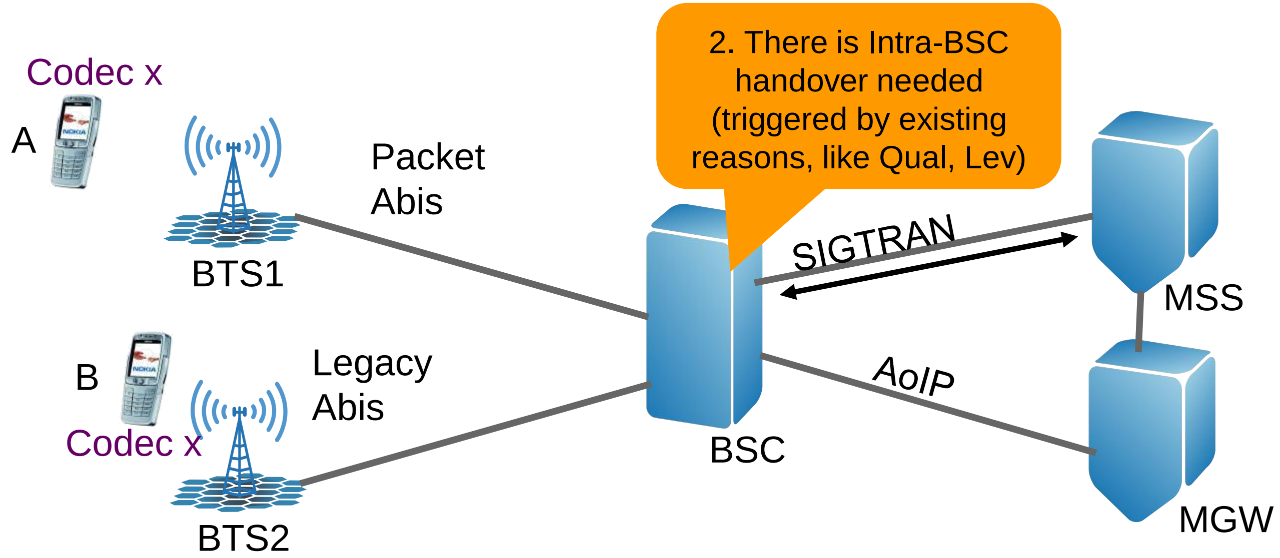

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Realization

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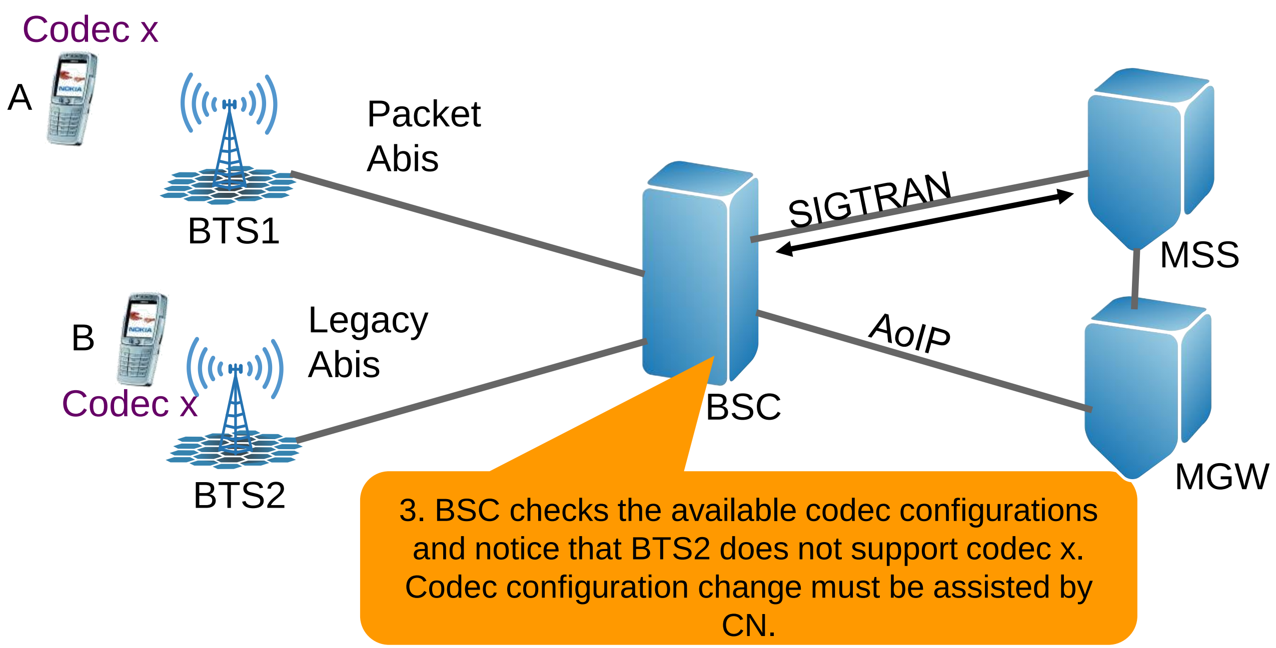

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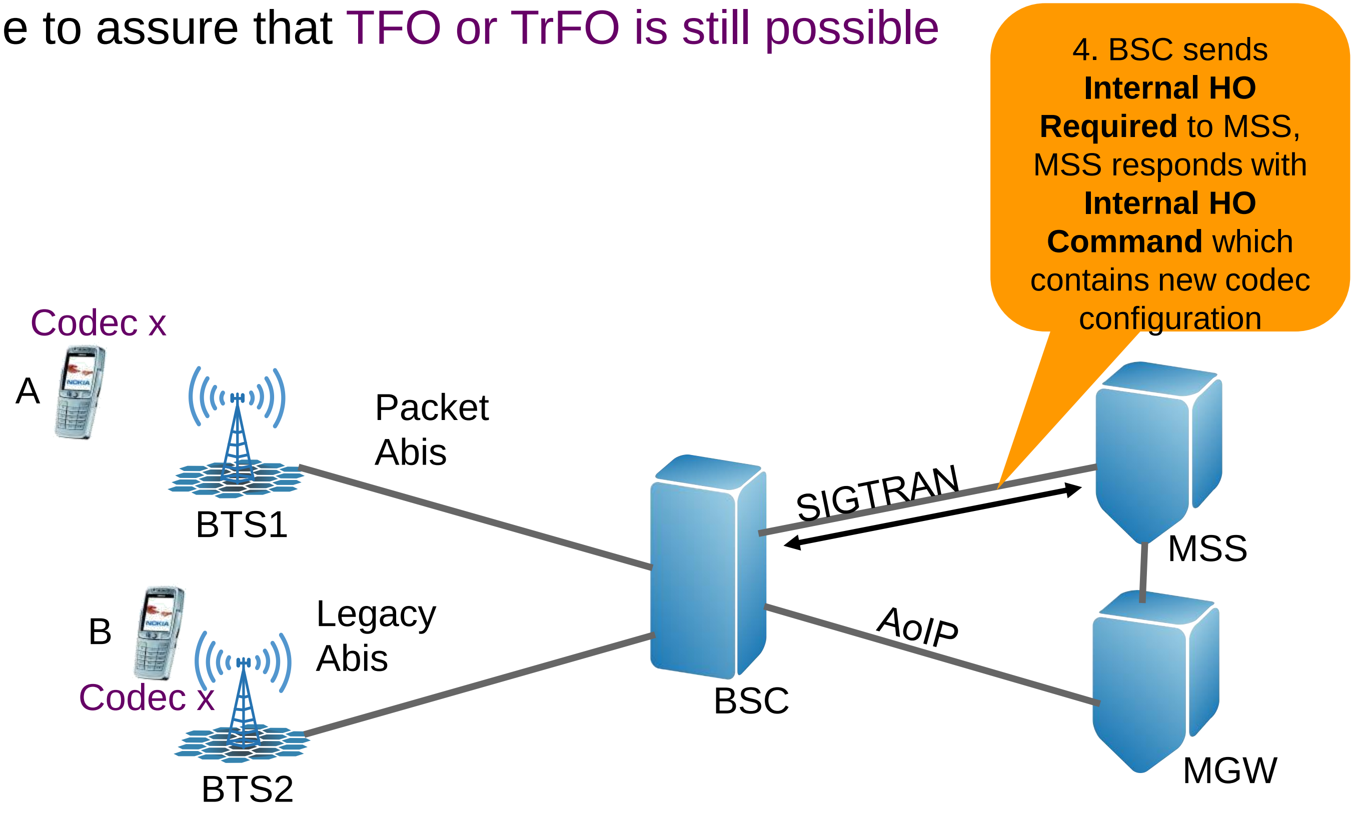

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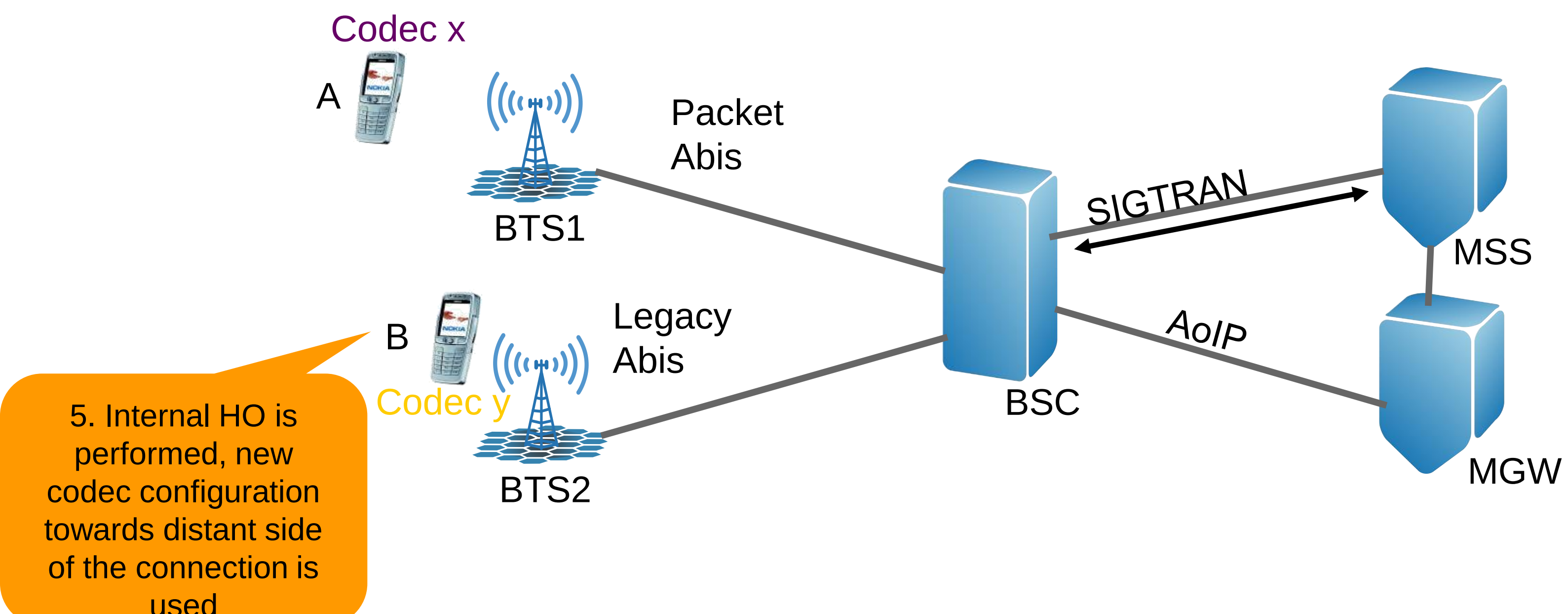

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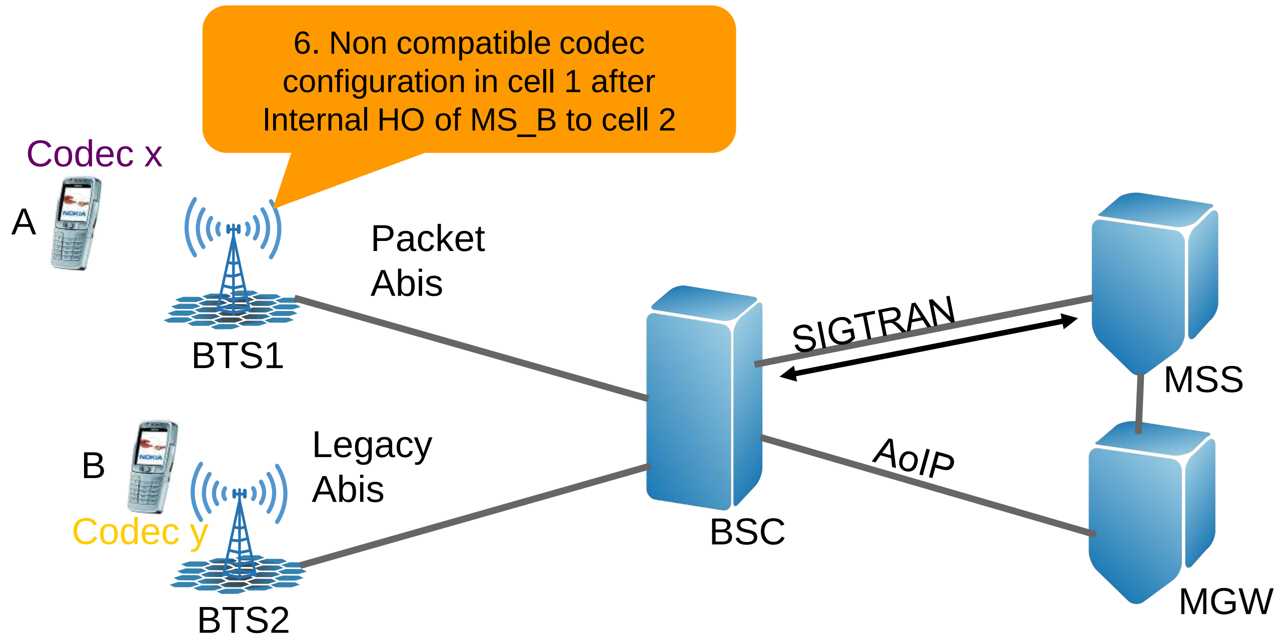

Realization

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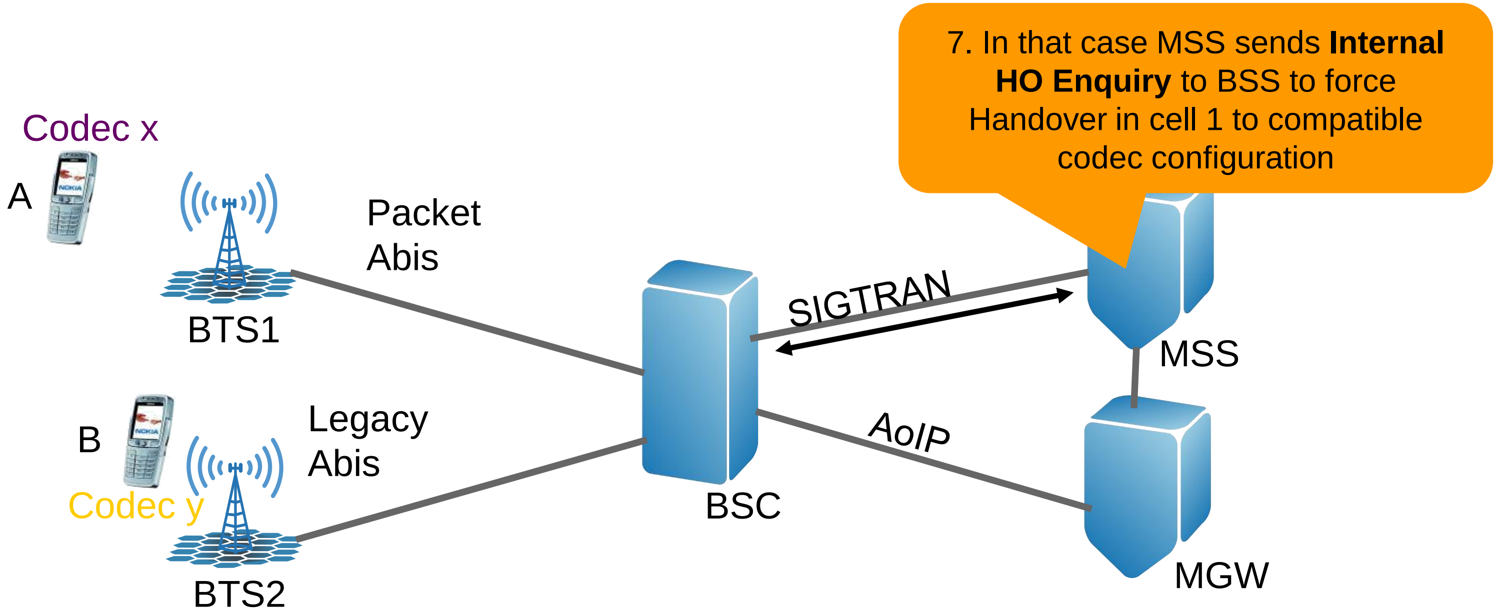

Realization

MSS controlled Intra-BSC Handovers 2/2

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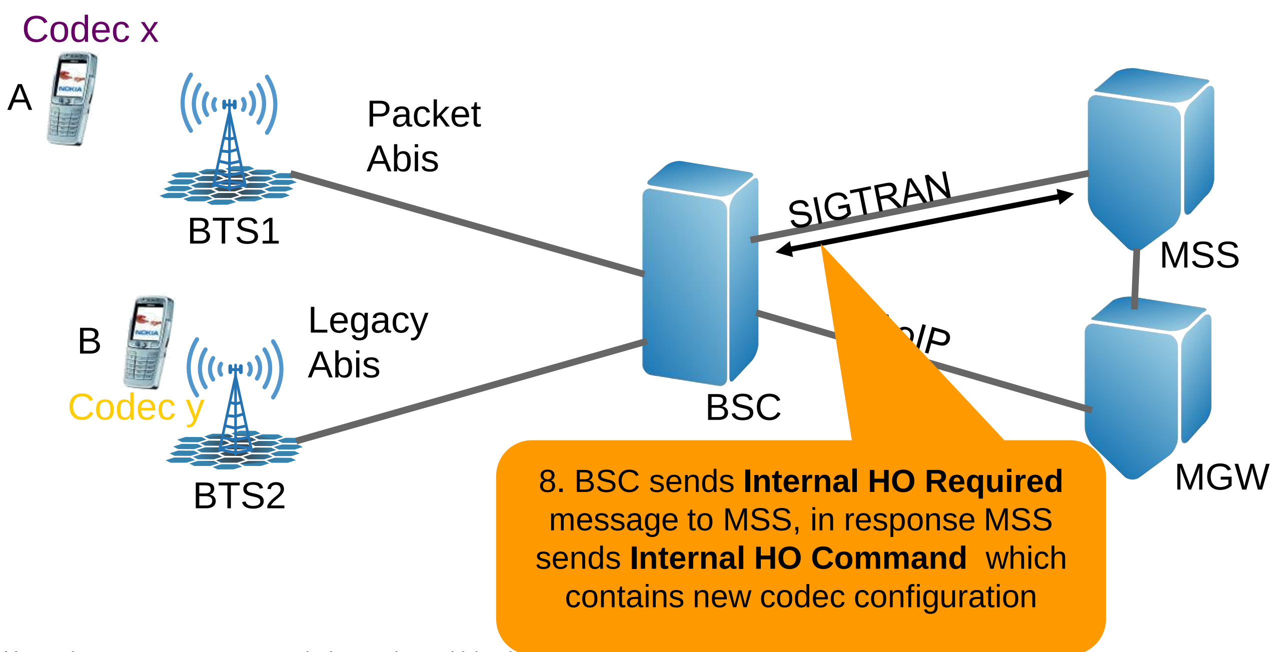

Realization

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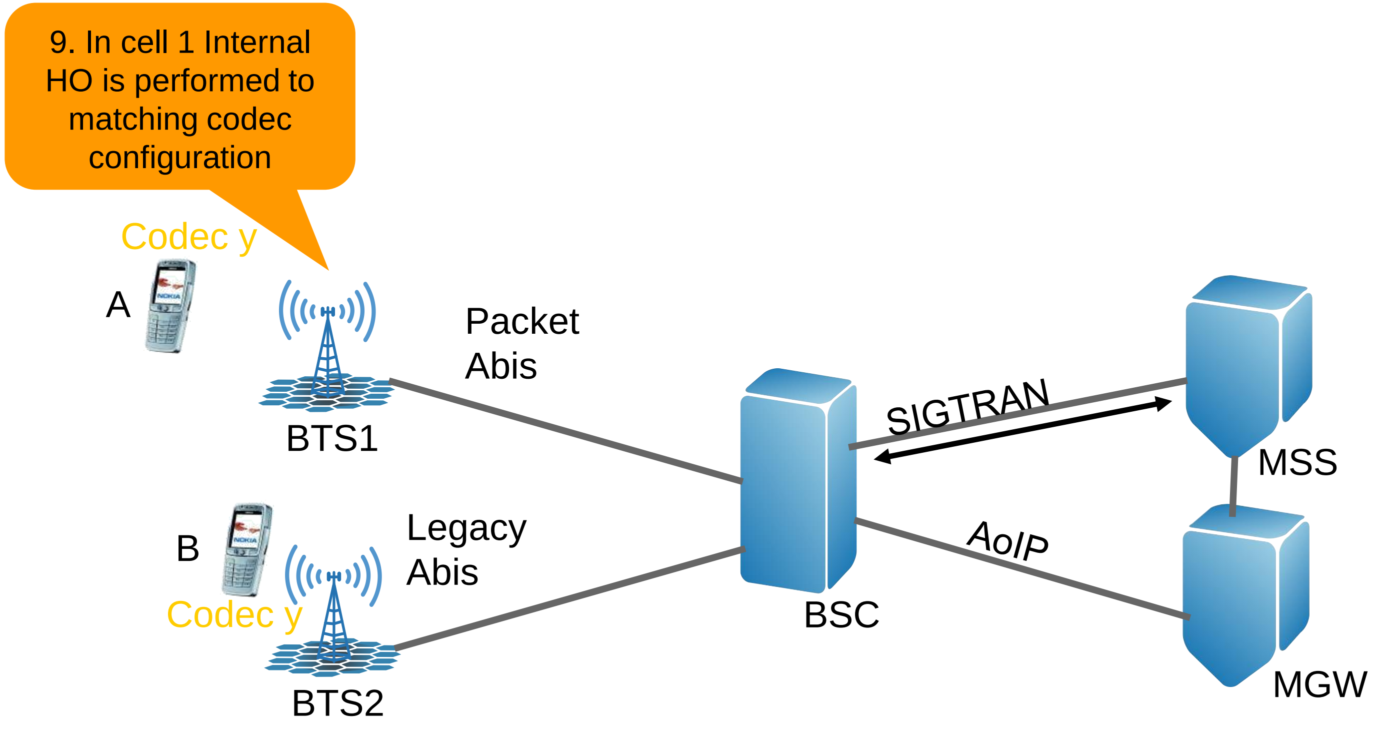
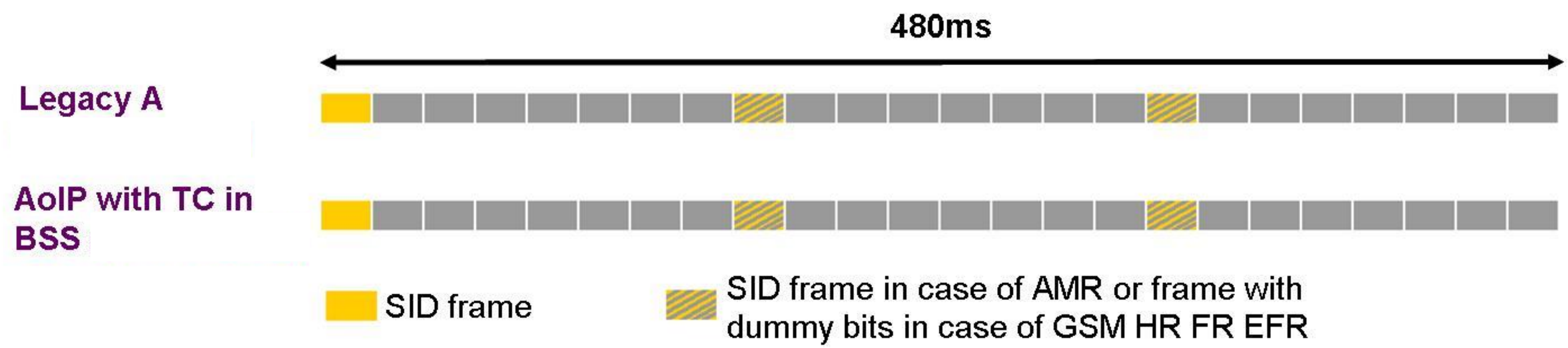


Realization

Silence suppression

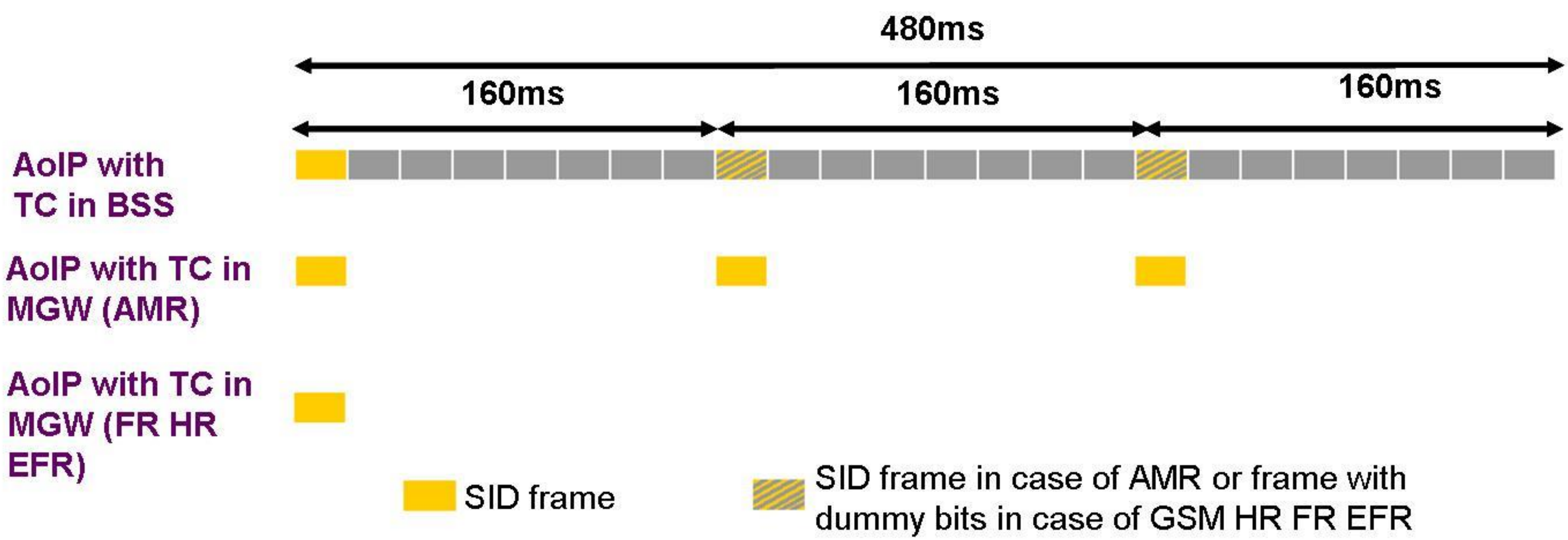
- Silence suppression handling depends on used A interface realization:
  - In case of Legacy A
    - In case of Discontinuous transmission detection (DTX) Silence Description (SID) frame is send once every 480ms (which corresponds to 24 TRAU frames) in case of GSM HR FR EFR and 160ms in case of AMR codecs
    - Remaining frames in that period are filled with dummy bits and transmitted across A interface
  - In case of AoIP with TC in BSS
    - Similar to Legacy A case transmission occurs all the time, SID frame is send every 160ms (AMR codecs) or 480ms (GSM FR HR EFR codecs), remaining frames in that period are filled with dummy bits and transmitted across A interface



Realization

Silence suppression

– In case of AoIP with TC in MGW
  - When DTX is detected one SID frame is send every:
    - 160ms in case of AMR codecs
    - 480ms in case of GSM FR HR EFR codecs
  - frames with dummy bits aren’t transmitted



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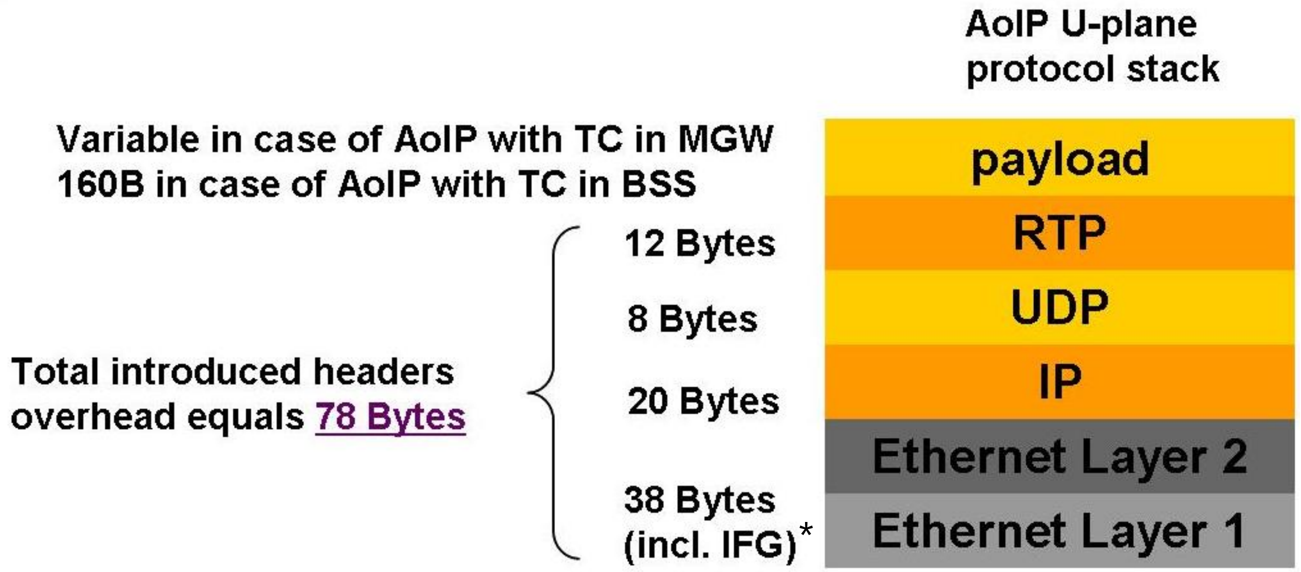
Realization

PSN realization

- AoIP feature introduces **completely different U-Plane traffic handling than in previous releases**
- All incoming frames are formed into packets and sent across PSN network (for more information about IP packet structure please refer to Packet Abis, M. Grygiel)
- There is **no header compression or CS multiplexing** on A interface
- That transmission concept requires completely new protocols stack

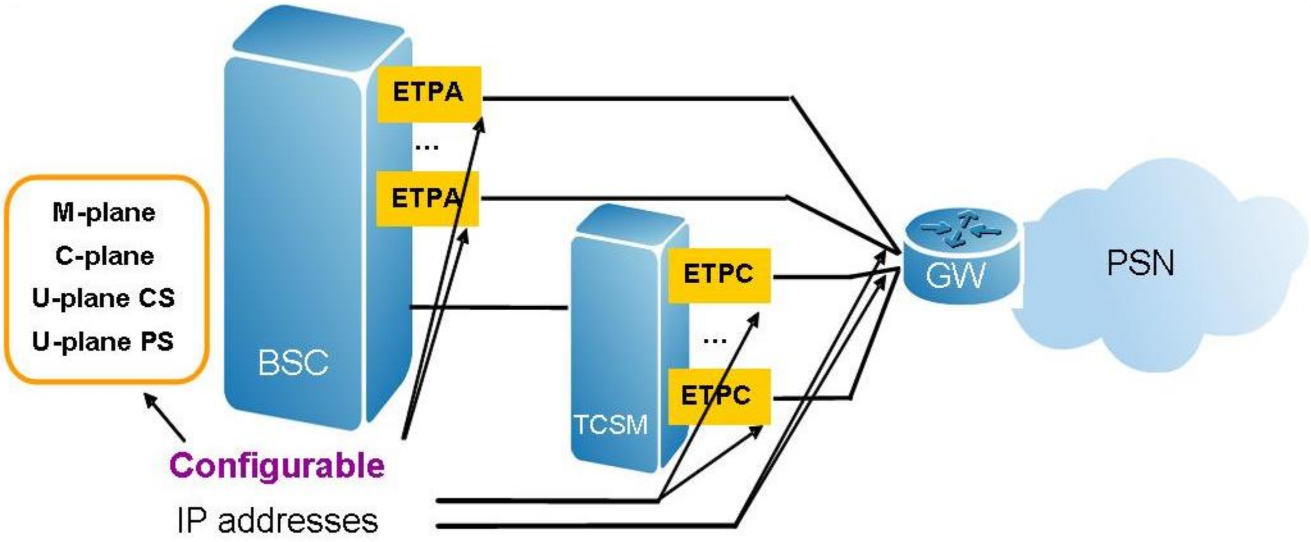


* Interframe Gap - idle period between transmission of Ethernet frames (12B)

Realization

Addressing basics

- Because Network Elements (NE) are connected to packet switched network (there is no direct connections between them as in legacy A realization) each of them needs proper addressing to be able to communicate across the network
- In RG20 there is used static addressing based on IPv4 (IPv6 is not supported)
- That means that **every single NE interface** that will use PSN **must be manually configured with IP address and subnet mask**
- A decision must be made on the basis of each network if private or public addresses will be in use



Dimensioning and planning aspects

Dimensioning and planning aspects

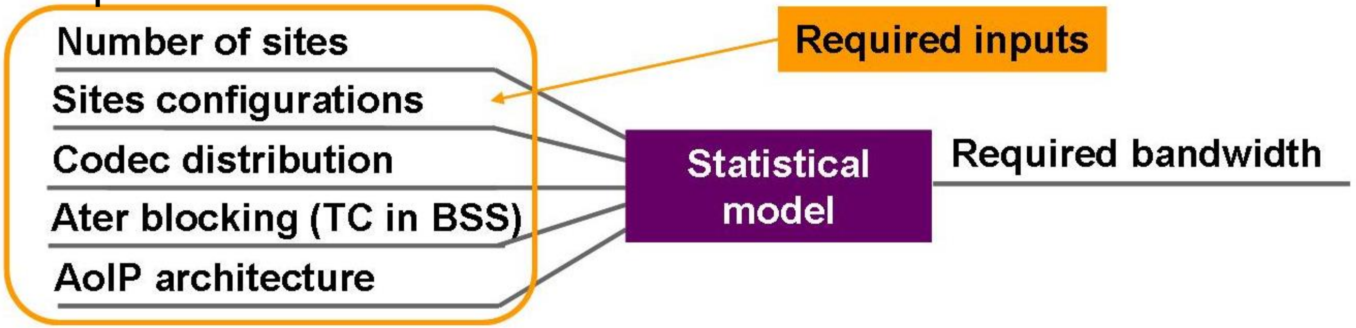
Legacy A dimensioning procedure

- Legacy A realization dimensioning procedure aims at estimation of the number of required Ater and A interface PCM lines
- The number of required Ater PCM lines depends on the number of required TCH and SS7 channels on Ater interface:
  - TCH dimensioning on Ater interface is done based on the total amount of Erlangs switched by the BSC and required blocking rate on Ater interface
  - SS7 dimensioning is based on the number of subscribers served by the BSC and subscribers behavior described by means of traffic model
- When the number of TCH and SS7 channels is known it is possible to estimate the required number of Ater PCM lines
- The last step is to calculate the required number of PCM lines on A interface (#PCM lines on A-interface = 4 * #PCM lines on Ater)

Dimensioning and planning aspects

AoIP dimensioning procedure

- AoIP dimensioning procedure is completely different from the legacy A procedure and aims at finding **bandwidth sufficient to transport** through backhaul the **offered traffic for given network configuration (TRX site configuration and site count), traffic load and codec distribution**
- Signaling traffic has dedicated interface on Ater or/and A and is not transmitted together with U-plane



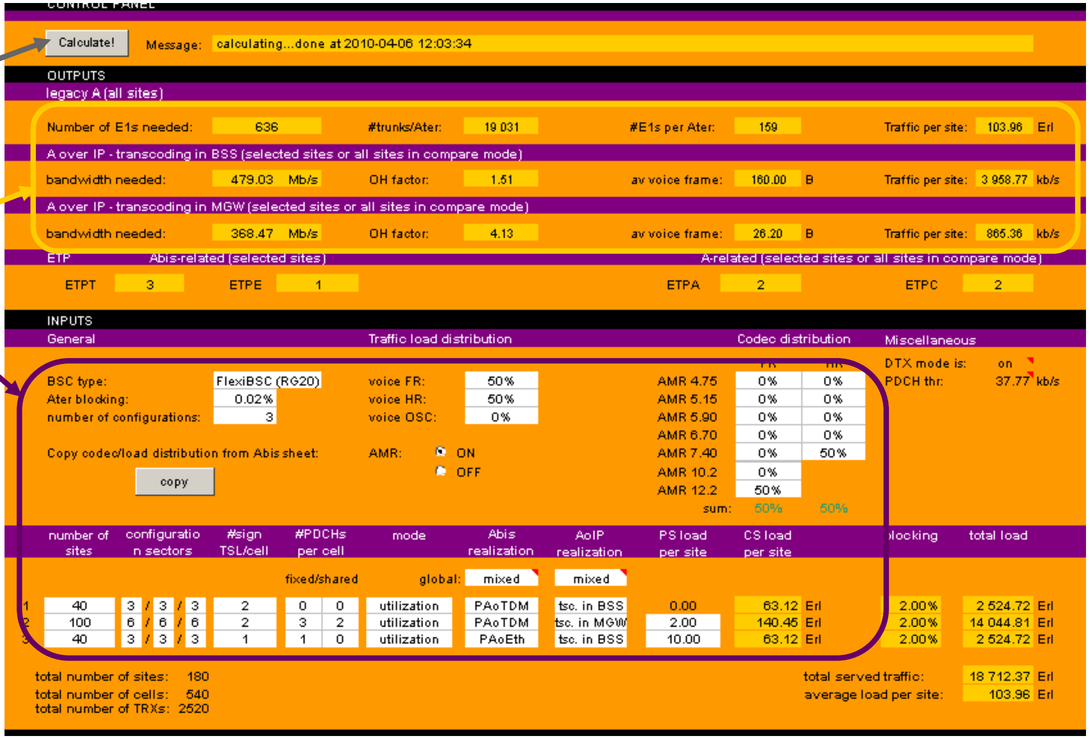
- To speed up and simplify dimensioning procedure there is AoIP bandwidth estimation tool available
- It allows to estimate required bandwidth needed on A interface
- In case of AoIP the tool uses statistical modelling based on ErlangC formula
- Statistics-based modeling is needed to model call arrival rate and find trade-off between delay and bandwidth

Dimensioning and planning aspects

AoIP calculator

- The Bandwidth estimation tool is stored on [IMS]
- AoIP calculator allows to estimate required bandwidth on both AoIP realizations and compare it with bandwidth consumption in Legacy A case

- Tool workflow:
  - enter **Inputs**
  - press **Calculate**
  - read-off **required A interface bandwidth**



Dimensioning and planning aspects

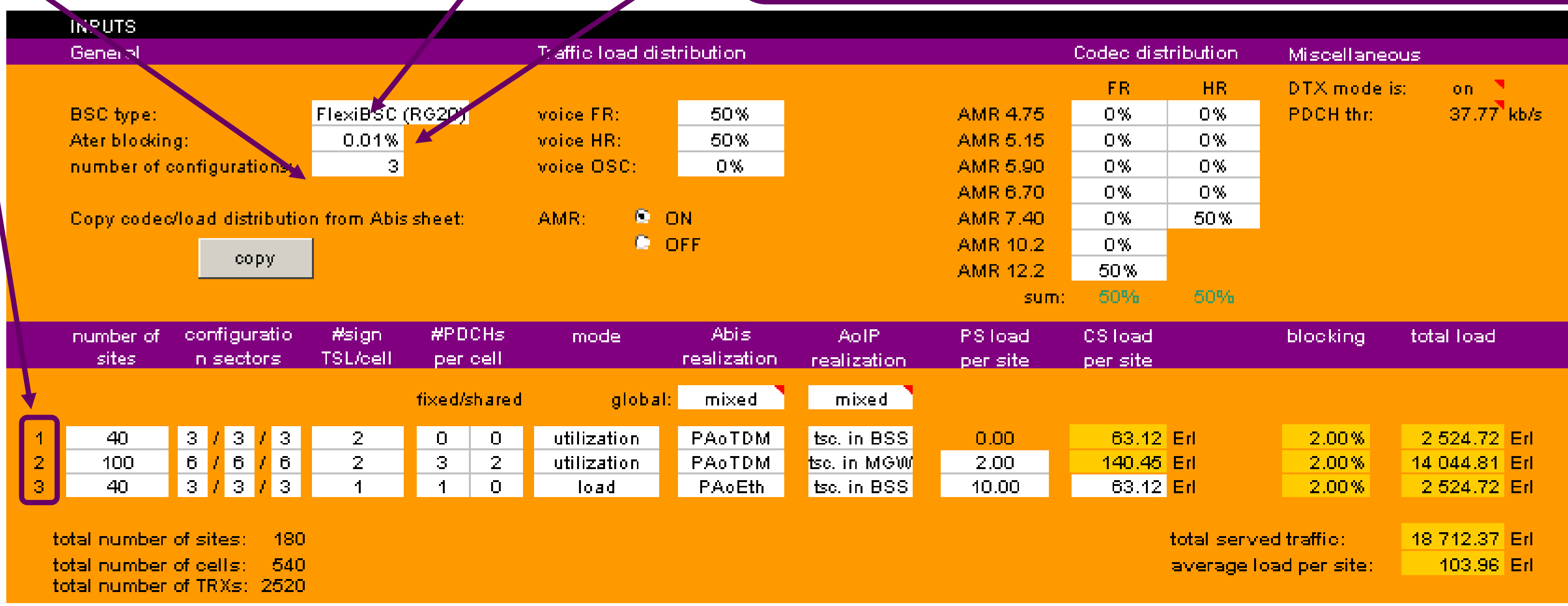
AoIP calculator – required inputs 1/4

- Required inputs are grouped in section Inputs:

Number of configurations (max 12) defines number of possible site configurations (from 1/1/1 to 16/16/16) that user can define

BSC type defines HW requirements related to specific BSC type(Max no. of TRXs, cells, sites and Erlangs)

Ater blocking defines blocking rate on Ater interface (usually 0.01%) in case of Legacy A and AoIP with TC in BSS calculations



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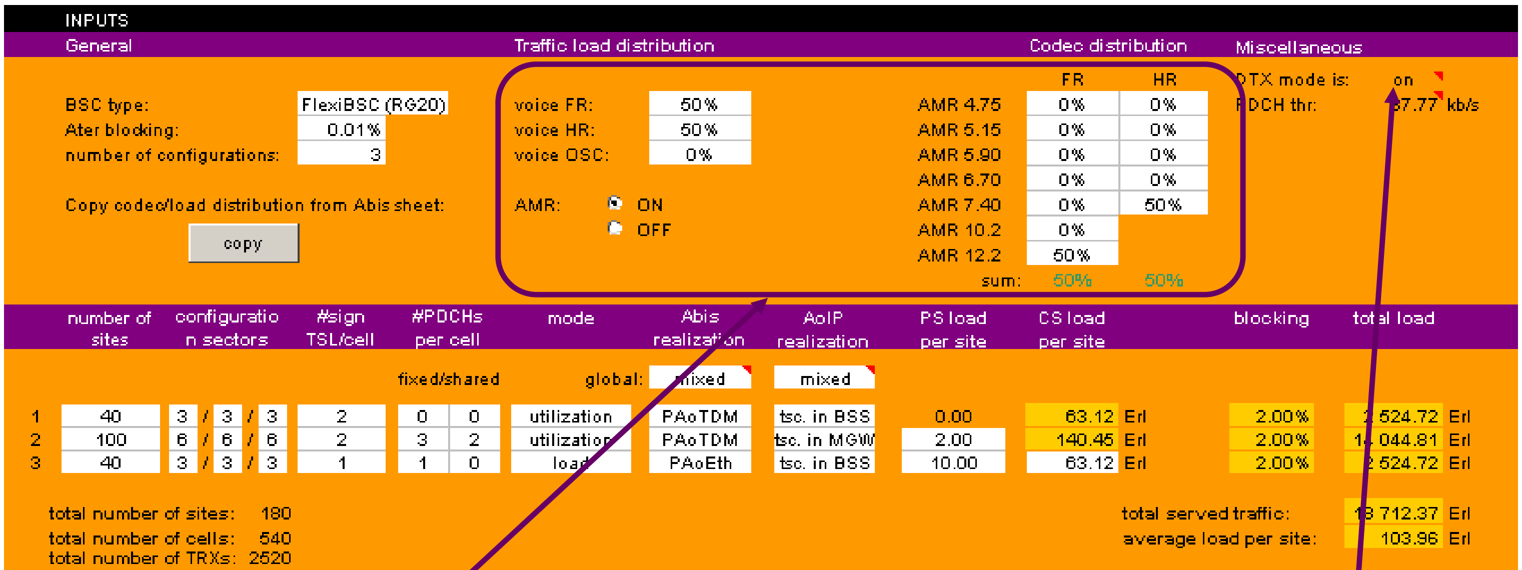
Dimensioning and planning aspects

AoIP calculator – required inputs 2/4

- Required inputs are grouped in section Inputs:



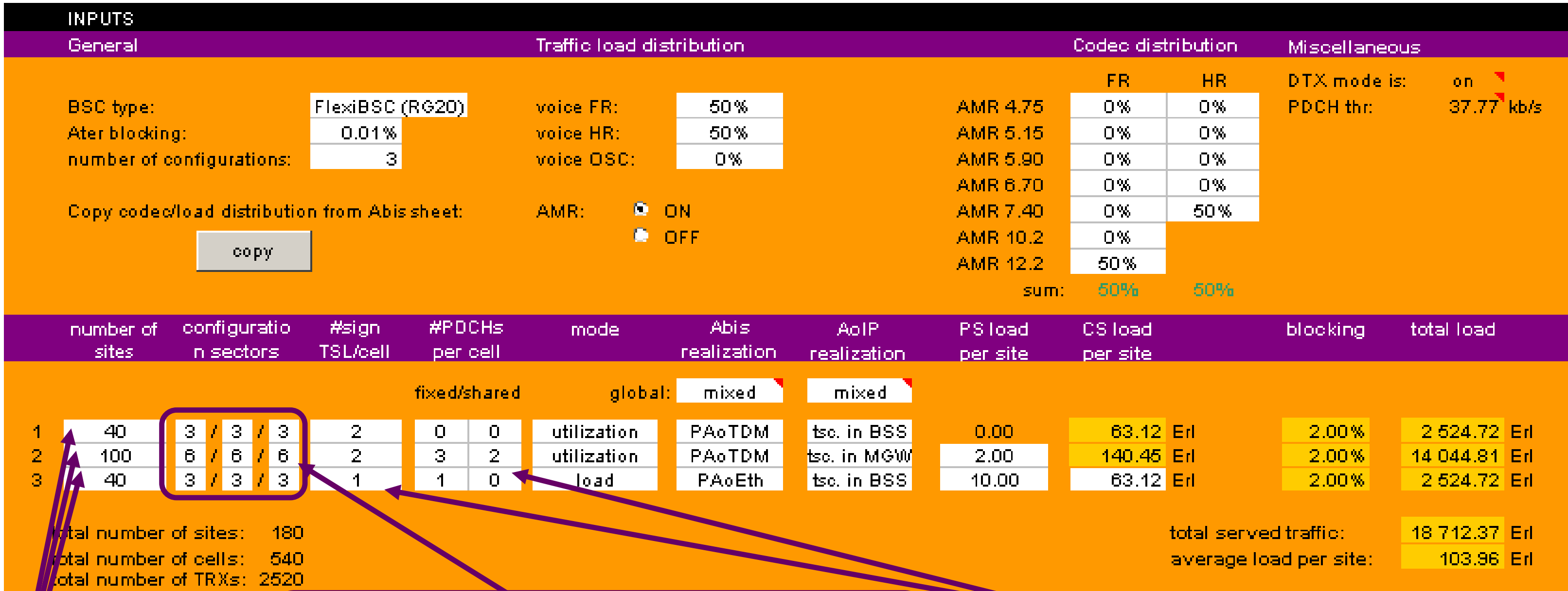
In this part user can define percentage share of voice codec used to bandwidth estimations. User can define only % share of codec type (AMR button “OFF”) or exact AMR codec distribution (AMR button “ON”) or can copy setting from Abis part with “copy” button

This field indicates that DTX is on. Change of the DTX state must be done in Packet Abis part of the tool (in complete view there is drop down lists that control setting of this field)

Dimensioning and planning aspects

AoIP calculator – required inputs 3/4

- Required inputs are grouped in section Inputs:



In this part user can define configuration of the sites

In this part user can define number of signaling timeslots and fixed or shared PDCH per configuration

In this part user can define number of sites in each configuration

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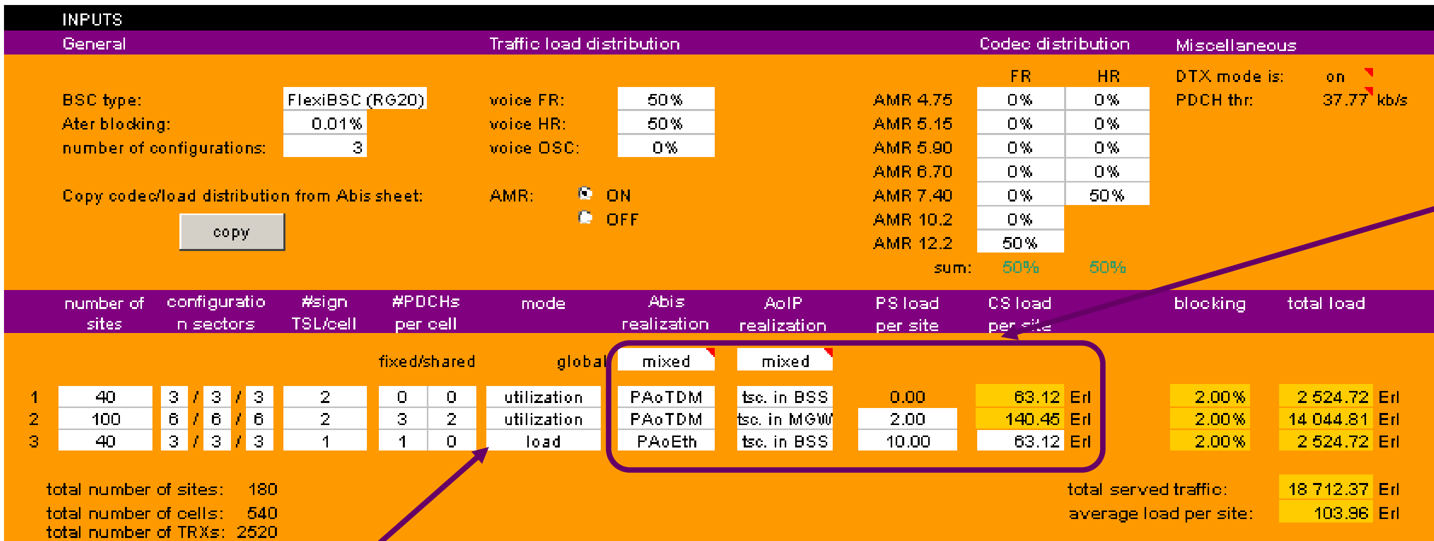
Dimensioning and planning aspects

AoIP calculator – required inputs 4/4

- Required inputs are grouped in section Inputs:



For description of this part of the tool please refer to slide 48

In this part user can choose mode of traffic definition:
load: user can enter the amount of Erlangs retrieved from the field
utilization: user can enter required percentage of FR/HR/OSC and tool will calculate the amount of FR/HR/OSC Erlangs that can be served by available RTSL assuming blocking rate of 2% (only hard blocking is to be met)

Dimensioning and planning aspects

AoIP calculator – results

- Calculation results are shown in section Outputs:

OUTPUTS											
legacy A											
Number of E1s needed:	292		#trunks/Ater:	8 702	#E1s per Ater:	73		Traffic per site:	42.24	Erl	
A over IP - transcoding in BSS											
bandwidth needed:	818.37	Mb/s	OH factor:	1.51	av voice frame:	160.00	B	Traffic per site:	2 703.46	kb/s	
A over IP - transcoding in MGW											
bandwidth needed:	236.17	Mb/s	OH factor:	3.60	av voice frame:	31.50	B	Traffic per site:	319.35	kb/s	

- There are 3 groups of results for each calculation:
  - **Required number of E1** needed for given inputs in case of Legacy A
  - **Required bandwidth needed** for given inputs in case of AoIP with TC in BSS
  - **Required bandwidth needed** for given inputs in case of AoIP with TC in MGW
- The displayed results denotes A interface required bandwidth (or number of E1 lines in case of legacy A) which is **needed** to transmit **offered CS traffic** with **given codec distribution, site configuration and site number for chosen configuration**
- **Results for legacy realizations are displayed for reference**
- **User choose required bandwidth depending on used scenario and ignore other results**

Dimensioning and planning aspects

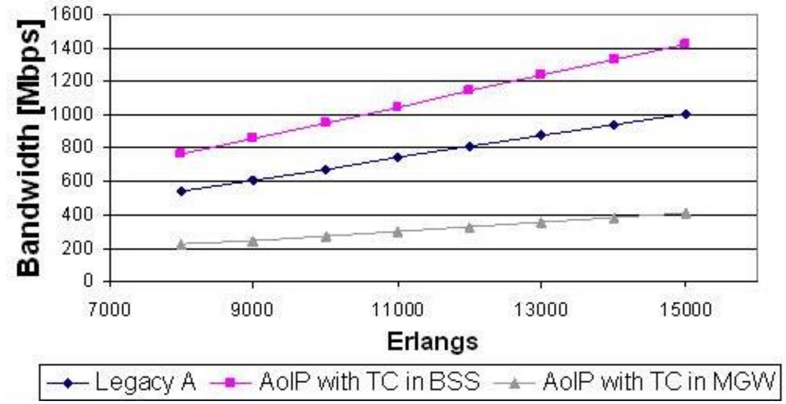
AoIP planning aspects 1/2

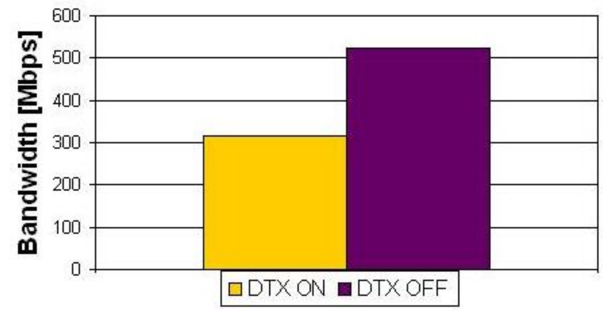
- Exemplary calculations from the Bandwidth estimation tool:
  - Number of sites per BSC – **200**,
  - Site configuration **3/3/3**,
  - no. of signalization TSL per BCCH TRX – **3**, **no PDCH channels (worst case)**
  - Ater blocking **0.01%**,
  - Codec distribution: **50% AMR 12.2**, **50% AMR 7.4 HR**
- Total traffic in Erlangs switched by BSC: **11848**
- In case of Legacy A **408 E1** lines are needed what gives **816Mbps**
- Required bandwidth on the A interface:
  - In case of AoIP with TC in BSS – **1.15Gbps**
  - In case of AoIP with TC in MGW – **315.07Mbps**
- In case of AoIP with TC in BSS required bandwidth is higher than in Legacy A case but usage of cost efficient IP/Eth transport more than compensates for the bandwidth consumption due to transport of headers introduced by AoIP
- AoIP with TC in MGW gives significant savings in terms of required bandwidth

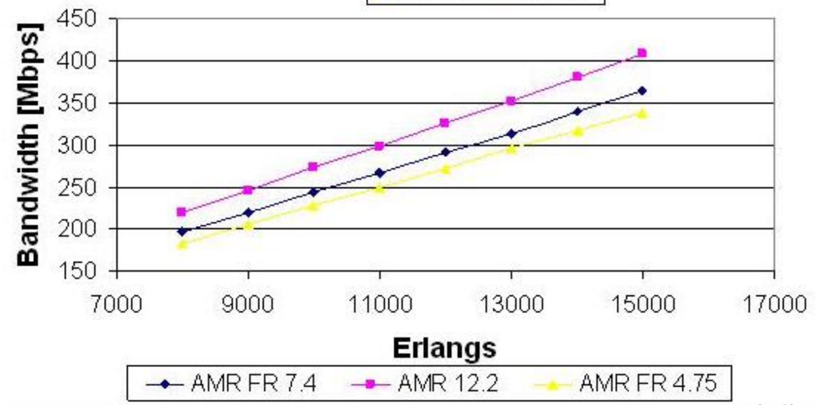
Dimensioning and planning aspects

AoIP planning aspects 2/2

- Main contributors to savings in case of AoIP with TC in MGW are:
  - Radio codecs transmission over A interface
    - Transmitted RTP payloads with radio codecs (ca. 30 Bytes) are much smaller than 160 Bytes long G.711 codecs
  - DTX transmission handling
    - In case of DTX ON bandwidth savings on A interface might be significant (e.g. and can reach 30 - 40% depending on voice activity factor and traffic distribution)
  - Codec distribution
    - Lower source bitrate codecs require less bandwidth







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Dimensioning and planning aspects

SS7 calculator

- SS7 calculator is used to estimate SIGTRAN bandwidth requirements
- Tool offers bandwidth estimation for any realization of SS7 signaling in RG20 including SIGTRAN which is mandatory for AoIP
- Tool options:

Microsoft Excel Worksheet

Input definition: either #Erlangs defined by user or calculated from traffic model

Amount of Erlangs switched in BSC

Traffic model configurable inputs

Required traditional SS7 bandwidth

C-plane realization

Required SIGTRAN bandwidth

Maximum link load in case of traditional SS7 signalization

input definition:	
enter directly (cell C17)	
User profile:	
#MOC	0.7
#MTC	0.3
#SMS	1
#LUP	2.1
#HO	0.8
holding time	17
#Erlangs per planning area:	10 000
Basic calculations:	
#SS7 load per user (bytes/hr)	708
#traffic load per user (mErl)	4.722222
#users per BSC	2 117 647
SS7 load per BSC (kbit/s)	3 331.8
SS7 realization:	
link type	SIGTRAN
max link load	40% not relevant for SIGTRAN
result:	
required bandwidth (Mbit/s)	14.04

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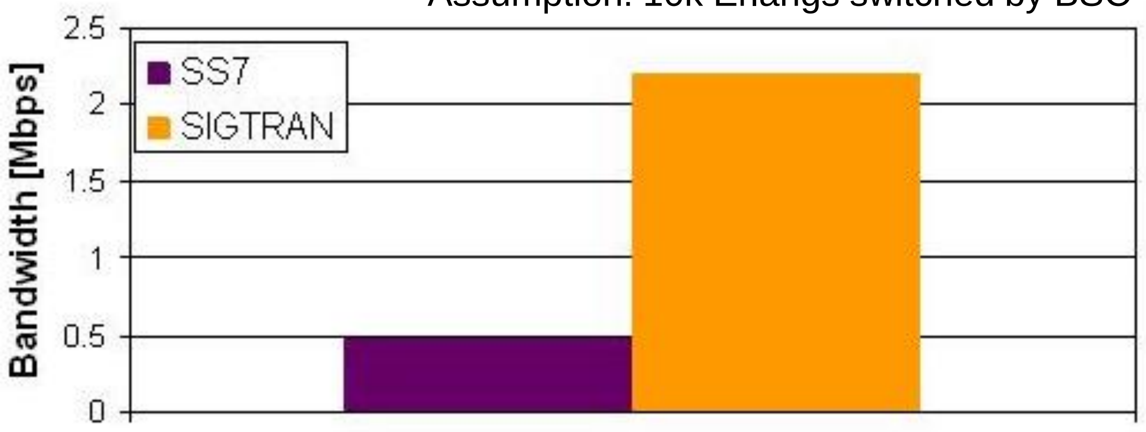
Dimensioning and planning aspects

SIGTRAN planning aspects 1/3

- Calculation for exemplary traffic model shows that SIGTRAN load is higher than for legacy SS7 over TDM signaling
- But similar as in AoIP case usage of **IP/Ethernet cost efficient transport compensates the bandwidth consumption introduced by SIGTRAN headers**
- Additionally SIGTRAN is not fixed to a certain bandwidth so there is no need to plan how many links will be needed for signaling

	Traffic model
#BHCA	1.0
MOC/BHCA	60%
#LU	2.0
#SMS-O	0.13
#SMS-T	0.07
#HO	0.5
Holding time	90sec

Assumption: 10k Erlangs switched by BSC



Dimensioning and planning aspects

SIGTRAN planning aspects 2/3

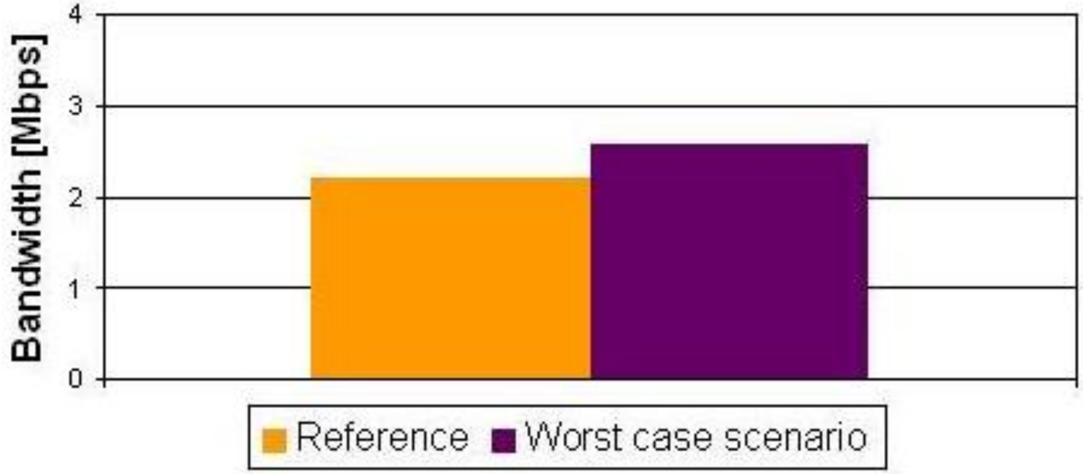
- AoIP feature requires MSS controlled Intra-BSC Handovers in some cases (see feature realization section)
- Intra-BSC HO caused by the codec changes to non matching codec configuration or A interface type change during the handover increase amount of required bandwidth for signaling because they require exchange of new messages what puts additional load on C-plane
- In **worst case** the #HO in used traffic model can be doubled (assuming that all HO require codec type, source bitrate or/and A-if type changes)

	Traffic model
#BHCA	1.0
MOC/BHCA	60%
#LU	2.0
#SMS-O	0.13
#SMS-T	0.07
#HO	**0.5 -> 1.0**
Holding time	90sec

Dimensioning and planning aspects

SIGTRAN planning aspects 3/3

- For traffic model given on previous slide SIGTRAN load from SS7 calculator increased for ~15%



- To monitor SIGTRAN load counters introduced in **661 M3UA ASSOCIATION SET MEASUREMENT** (BSS21170 Statistics for SIGTRAN feature) can be used
- Especially counters:
  - 661011 / M3UA SCTP OCTETS RECEIVED (DL_SIGTRAN_SCTP _BIT_RATE = M3UA_SCTP_OCTETS_RECEIVED * 8 / GRANURALITY (in seconds)
  - 661012 / M3UA SCTP OCTETS SENT (UL_SIGTRAN_SCTP _BIT_RATE = M3UA_SCTP_OCTETS_SENT * 8 / GRANURALITY (in seconds)

  Can give overview about SIGTRAN bitrate on SCTP layer
- Additionally that new HO types does not increase significantly signaling load on Abis interface
- **These findings must be confirmed in test campaign**

Dimensioning and planning aspects

Synchronization impact on planning

- Particular synchronization methods put different load on backhaul

Method	Bandwidth consumption for default settings
ToP (IEEE1588v2)	~ 11.8 kbps
ACR (CESoPSN)	~ 212 kbps
SyncEth	none (dedicated Eth link is used)

- In case of AoIP feature bandwidth used for synchronization purposes is negligible (~200kbps comparing to at least 20-30 and more Mbps)
- For details concerning please refer to Packet Abis Synchronization, A. Maciolek

Dimensioning and planning aspects

ETP dimensioning 1/2

- **BSC/TCSM configuration** possibilities:
  - BSC:
    - 2000/FlexiBSC: 6+6 ETP for Packet Abis (ETPT+ETPE) + 4 ETP for AoIP (ETPA)
    - 1000: 2+2 ETP for Packet Abis (ETPT+ETPE) + 2 ETP for AoIP (ETPA)
  - TCSM3i: 5+5 ETP for AoIP (ETPC)
- **ETP capacity:**
  - ~8000 TCH in case of ETPA
  - 3840 TCH in case of ETPC
- **Exemplary calculation:**
  - Given:
    - 100 sites with 37 TCH/cell and 80 Erl/site
    - 50 sites with 22 TCH/cell and 40 Erl/site
  - Calculations:
    - 100 x 80 Erl + 50 x 40 Erl = 10 kErl/BSC => 10,170 TCH (0.1% blocking)
    - 2 ETPA needed in case of AoIP with TC in MGW
    - 3 ETPC needed in case of AoIP with TC in BSS

Dimensioning and planning aspects

ETP dimensioning 2/2

- ETP planning is fully supported by **BW estimation tool**
- Tool workflow:
  - Define network configuration:
    - number of sites with given configuration
    - for each site configuration all relevant inputs must be given (cf. Slide129 for details), e.g. #TRX/sector, #of signaling RTSL, CS and PS traffic, CS/PS codec distribution
    - for each site configuration also two parameters which are ETP planning specific must be given, i.e. **Abis realization** and **AoIP realization** (see callouts below)

For each site type user can indicate whether Legacy Abis (site is excluded from ETP dimensioning) or Packet Abis over TDM or Packet Abis over Ethernet is used. Field **global** can speed up the setting for large networks: “mixed” means that Abis realization varies among sites while any specific realization (again “Legacy Abis”, “PAoTDM”, “PAoEth”) is chosen is loaded automatically in each and every site

For each site type user can indicate whether AoIP with TC in BSS or in MGW is used (note: tool assumes that AoIP is always in use; if this is not the case then the results related to ETP-A can be just ignored). Field **global** can speed up the setting for large networks: “mixed” means that AoIP realization varies among sites while any specific realization (again “tsc in BSS”, “tsc in MGW”) if chosen is loaded automatically in each and every site; value “compare” gives at once the results as if all sites would use AoIP with TC in BSS and in MGW for comparison purposes

	number of sites	configuration sectors 1/2/3	#sign TSL/cell	#PDCHs per cell		mode	Abis realization	AoIP realization	PS load per site [MB/h]	CS load per site
				fixed/shared		global:	mixed	compare		
1	100	5 / 5 / 5	2	2	2	load	PAoTDM		5,00	100,00 Erl
2	50	3 / 3 / 3	2	2	2	load	PAoEth		5,00	50,00 Erl

  - Then press **calculate** and read-off the results:

ETP	Abis-related (selected sites)			A-related (selected sites or all sites in compare mode)			
ETPT	2	ETPE	1	ETPA	2	ETPC	4

AoIP feature benefits

Benefits

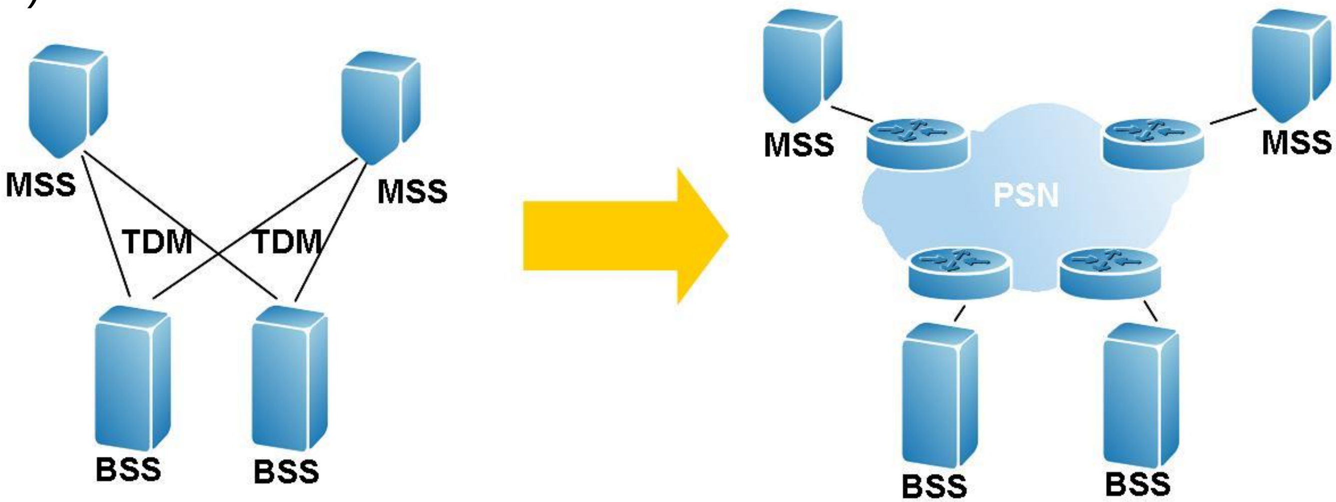
A over IP benefits

- **Usage of IP connectivity over A interface gives following benefits:**
  - Cost savings in transport, because IP transport is most cost effective
  - Savings in maintenance (easier configuration – same backhaul network for all kind of traffic)
  - Lease of PSN networks is more cost effective than TDM networks
  - Flexible user plane routing
  - Flexible network design
  - Better network resiliency (IP network can self reconfigure after failure one of the routes)
  - Backbone can be shared between different traffic planes
  - Backbone can be shared with different services and even operators
  - Better speech quality and in case of AoIP with TC in MGW – bandwidth savings (TrFO and smaller data payloads)

Benefits

A over IP benefits

- **Benefits for Multipoint – A (BSS12 feature):**
  - Easier configuration:
    - In PSN there is no need to connect each MSC with BSC with dedicated physical TDM connection (as it was earlier)
    - PSN can easily route packets to each entity having proper configuration and access to PSN network
  - Connectivity between NEs is most cost effective with usage of IP technology
  - Improved resiliency of backbone network (IP cloud can reconfigure routes in case of failures)



Configuration management

Configuration management

New Parameters 1/12

- PAFILE parameters

Parameter	Description
T25 **object:** BSC **range:** 5-30s **default:** 20s	*Internal Handover required periodicity. This parameter is used to set time which BSC can wait for response from MSC for INTERNAL HANDOVER REQUIRED MESSAGE. If time defined in this parameter exceed and retry is now allowed (T25_rep), Internal HO procedure is aborted. PAFILE parameter*
T25_rep **object:** BSC **range:** 1-5 **default:** 2	*Number of T25 repetitions. See explanation above. PAFILE parameter*
Trbss **object:** BSC **range:** 12-15 **default:** 12	*Time to receipt of RESET IP RESOURCE ACKNOWLEDGE at the BSS. If time defined by this parameter expires RESET IP RESOURCE procedure fails. PAFILE parameter*

Configuration management

New Parameters 2/12

- BSC parameters - DiffServ

Parameter	Description
A-If-CS-U-plane DSCP PHB TC in MGW (AIfCsUplaneDscpPhbTcInMgw) **object:** BSC **range:** (0 – 63) **default:** 34 (AF41) **MML command:** EE1, EEO **MML abbreviated name:** ADSCPM	*DSCP PHB for CS U-Plane traffic class on A interface when TC is in MGW. The setting of the parameter is mapped to a 8-bit ToS field in IP packet header. 6 least significant bits are used for encoding of DiffServ Code Points (DSCP).* *Mapping between the QoS classes, the AoIP traffic Classes and the DiffServ Code Points (DSCP) Per Hop Behaviors (PHB) is needed because AoIP must threat each traffic class in same way as Packet Abis to assure proper QoS at all interfaces (for more info about QoS please refer to Packet Abis NEI, M. Grygiel, slide 99* *Note: Range – DSCP value shall be give either binary string or in hexadecimal. Step: 0 (BE), 10 (AF11), 18 (AF21), 26 (AF31), 34 (AF41), 46 (EF)*
A-If-CS-U-plane DSCP PHB TC in BSS (AIfCsUplaneDscpPhbTcInBss) **object:** BSC **range:** (0 – 63) **default:** 46 (EF) **MML command:** EE1, EEO **MML abbreviated name:** ADSCPB	*DSCP PHB for CS U-Plane traffic class on A interface when TC is in BSS. For description see parameter above.*

Configuration management

New Parameters 3/12

- BSC parameters – A-If Traffic Scheduling Parameters

Parameter	Description
A-If AF4 WFQ weight (AIfAf4WfqWeight) **object:** BSC **range:** 1 – 10000, step 1 **default:** 5000 **MML command:** EE1, EEO **MML abbreviated name:** AAF4WFQ	*With this parameter user define the Weighted Fair Queuing (WFQ) weight assigned to the Assured Forwarding Class 4 (AF4) scheduling queue in A interface.* *This is one of the traffic scheduling parameters (see Packet Abis NEI, slide 99 for more details) and it has meaning only if same interface terminates several planes with different QoS. Parameter is used in used in ETP.*
A-If AF3 WFQ weight (AIfAf3WfqWeight) **object:** BSC **range:** 1 – 10000, step 1 **default:** 5000 **MML command:** EE1, EEO **MML abbreviated name:** AAF3WFQ	*With this parameter user define the Weighted Fair Queuing (WFQ) weight assigned to the Assured Forwarding Class 3 (AF3) scheduling queue in A interface.* *This is one of the traffic scheduling parameters (see Packet Abis NEI, slide 99 for more details) and it has meaning only if same interface terminates several planes with different QoS. Parameter is used in used in ETP.*

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Configuration management

New Parameters 4/12

- BSC parameters – A-If Traffic Scheduling Parameters

Parameter	Description
A-If AF2 WFQ weight (AIfAf4WfqWeight) **object:** BSC **range:** 1 – 10000, step 1 **default:** 5000 **MML command:** EE1, EEO **MML abbreviated name:** AAF2WFQ	*With this parameter user define the Weighted Fair Queuing (WFQ) weight assigned to the Assured Forwarding Class 2 (AF2) scheduling queue in A interface.* *This is one of the traffic scheduling parameters (see Packet Abis NEI, slide 99 for more details) and it has meaning only if same interface terminates several planes with different QoS. Parameter is used in used in ETP.*
A-If AF1 WFQ weight (AIfAf3WfqWeight) **object:** BSC **range:** 1 – 10000, step 1 **default:** 5000 **MML command:** EE1, EEO **MML abbreviated name:** AAF1WFQ	*With this parameter user define the Weighted Fair Queuing (WFQ) weight assigned to the Assured Forwarding Class 1 (AF1) scheduling queue in A interface.* *This is one of the traffic scheduling parameters (see Packet Abis NEI, slide 99 for more details) and it has meaning only if same interface terminates several planes with different QoS. Parameter is used in used in ETP.*

Configuration management

New Parameters 5/12

- BSC parameters – A-If Traffic Scheduling Parameters

Parameter	Description
A-If BE WFQ weight (AIfBeWfqWeight) **object:** BSC **range:** 1 – 10000, step 1 **default:** 5000 **MML command:** EE1, EEO **MML abbreviated name:** ABEWFQ	*With this parameter user define the Weighted Fair Queuing (WFQ) weight assigned to the Assured Forwarding Class 2 (AF2) scheduling queue in A interface.* *This is one of the traffic scheduling parameters (see Packet Abis NEI, slide 99 for more details) and it has meaning only if same interface terminates several planes with different QoS. Parameter is used in used in ETP.*

Configuration management

New Parameters 6/12

- BSC parameters – V-LAN priority mapping

Parameter	Description
VLAN priority BE (vpBe) **object:** BSC **range:** 0 – 6, step 1 **default:** 0 **MML command:** EEY, EEO, EE1 **MML abbreviated name:** VPBE	*This parameter defines VLAN priority bits to DSCP PHB BE. Priority definition for the external interface (BSC-BTS) is defined with user parameter. The setting of the parameter is mapped to a 12-bit VID field in Ethernet header specifying the VLAN which the frame belongs to. There are two specific values of VID reserved for management of VLAN tagging:* *- a value of 0 means that the frame doesn't belong to any VLAN* *- a value of FFF(hex)=4095 is reserved for implementation use* *All other values may be used as VLAN identifiers, allowing up to 2^{12} – 2 = 4094 VLAN codes. However, only 2 of them can be used in RG20.* *(see Packet Abis NEi, slide 100 and 104 for more details)*
VLAN priority AF1 (vpAf1) **object:** BSC **range:** 0 – 6, step 1 **default:** 1 **MML command:** EEY, EEO, EE1 **MML abbreviated name:** VPAF1	*This parameter defines VLAN priority bits to DSCP PHB AF1. For description see parameter above.*

Configuration management

New Parameters 7/12

- BSC parameters – V-LAN priority mapping

Parameter	Description
VLAN priority AF2 (vpAf2) **object:** BSC **range:** 0 – 6, step 1 **default:** 3 **MML command:** EEY, EEO, EE1 **MML abbreviated name:** VPAF2	*This parameter defines VLAN priority bits to DSCP PHB AF2. For description see parameter on previous slide.*
VLAN priority AF3 (vpAf3) **object:** BSC **range:** 0 – 6, step 1 **default:** 4 **MML command:** EEY, EEO, EE1 **MML abbreviated name:** VPAF3	*This parameter defines VLAN priority bits to DSCP PHB AF3. For description see parameter on previous slide.*

Configuration management

New Parameters 8/12

- BSC parameters – V-LAN priority mapping

Parameter	Description
VLAN priority AF4 (vpAf4) **object:** BSC **range:** 0 – 6, step 1 **default:** 5 **MML command:** EEY, EEO, EE1 **MML abbreviated name:** VPAF3	*This parameter defines VLAN priority bits to DSCP PHB AF4. For description see parameter on previous slide.*
VLAN priority EF (vpEf) **object:** BSC **range:** 0 – 6, step 1 **default:** 6 **MML command:** EEY, EEO, EE1 **MML abbreviated name:** VPEF	*This parameter defines VLAN priority bits to DSCP PHB EF. For description see parameter on previous slide.*

Configuration management

New Parameters 9/12

- PRFILE Parameters

Parameter	Description
Base RTP port ETPA (BASE_RTP_PORT_ETPA) **object:** ETPA **range:** 49152…49535 **default:** 49152	*This parameter defines the base UDP port and thus that the range of all available UDP ports that are negotiated (UDP port values of ETPA are not fixed) during Call Setup and Handovers (when required) for CS U-Plane RTP traffic when AoIP with TC in MGW is used. Only even numbers are allowed as value for this parameter.*
Base RTP port ETPC (BASE_RTP_PORT_ETPC) **object:** ETPC **range:** 57344…57535 **default:** 57344	*This parameter defines the base UDP and thus that the range of all available UDP ports that are used for CS U-Plane RTP traffic when AoIP with TC in BSS is used. Only even numbers are allowed as value for this parameter.* *In case of AoIP with TC in BSS there is used fixed mapping between Ater channel (PCM, TSL) to transport address of the ETPC (IP address and port number for RTP). In that way when BSC selects Ater-channel for the call, BSC knows the corresponding A-interface IP termination point in ETPC according to the equation:* ***UDP Port =((ETP_PCM_ID*32 + TSL)*2) + BASE_RTP_PORT_ ETPC***
CS Call Length (CS_CALL_LENGTH) **object:** ETPA **range:** 0 - 180 **default:** 180	*ETPA shall monitor call which are longer than this parameter for detecting hanging calls. The call duration is indicated in minutes.*

Configuration management

New Parameters 10/12

- PRFILE Parameters – PDV

Parameter	Description
Packet Delay Variation (PKT_DELAY_VAR) **object:** ETPC **range:** 0 – 50, step 1ms **default:** 0	*Due to characteristics of IP networks there is a delay variation (jitter) between packets, which can cause packet loss and decreased voice quality. In AoIP jitter buffer is used to compensate that variation.* *Jitter buffer is a shared memory area where incoming packets can be collected, temporarily stored, and sent to further processing (for more info about jitter in PSN please refer to Packet Abis Synchronization NEI)* *That buffering takes place in different NEs depending on used architecture:* ***TC in MGW case** – jitter buffer is located in BTS as BTS terminates IP connections* ***TC is in BSS case** – dejittering takes place in ETPC (locates in TCSM3i)* *ETPC has a static Jitter buffer for **each CS connection** which compensates jitter by delaying reconstruction of PCM frame from RTP packet. Usage of the buffer introduces known static delay (based on expected max PDV) instead of jitter effect* *The optimum **size of the buffer must be figured out** in test phase as it affects feature performance and user perceivable quality: if a packet arrives too late, i.e. has greater delay than expected max, then it will be discarded: discarding of an excessive number of packets (buffer underrun) will lead to call quality degradation and too large Jitter buffer (buffer overrun) will lead to conversational difficulty*

Configuration management

New Parameters 11/12

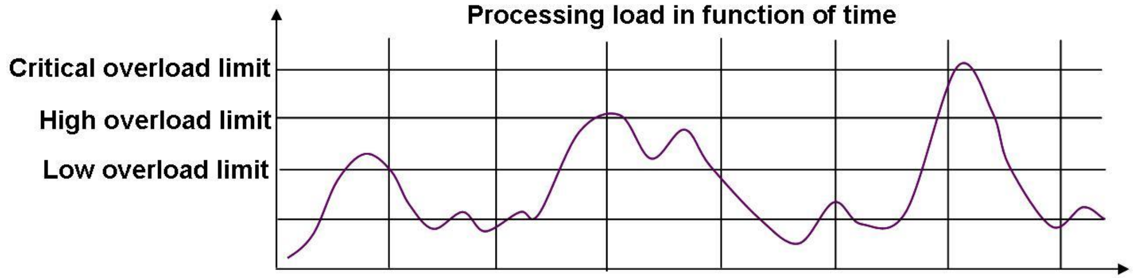
- PRFILE Parameters – ETP overload

Parameter	Description
LOW_OVERLOAD_LIMIT **range:** 0 – 100, step 1 **default:** 60	*This parameter is used by ETP to configure Low Overload Threshold parameter. If this threshold is exceeded ETPx unit limit non-essential services (like Logging, Message Monitoring and Statistic Collection) and reject the new call setups (except emergency calls). Value is percentage.* *New value will be provided to ETPA/C during ETPA/C restart and with AoIP related BSC parameter modification.*
HIGH_OVERLOAD_LIMIT **range:** 0 – 100, step 1 **default:** 70	*This parameter is used by ETP to configure High Overload Threshold parameter. If this threshold is exceeded ETPx unit reject all new call setups. Value is percentage.* *New value will be provided to ETPA/C during ETPA/C restart and with AoIP related BSC parameter modification.*
CRITICAL_OVERLOAD_LIMIT **range:** 0 – 100, step 1 **default:** 80	*This parameter is used by ETP to configure Critical Overload Threshold parameter. If this threshold is exceeded ETPx block all PS traffic from overloaded PCU – it is not a case in A over IP* *New value will be provided to ETPA/C during ETPA/C restart and with AoIP related BSC parameter modification.*

Configuration management

New Parameters 12/12

- PRFILE Parameters – ETP overload



Parameter	Description
LOAD_HYSTERESIS **range:** 0 – 10, step 1 **default:** 5	*This is used by ETP to configure Load Hysteresis parameter. This parameter determines percentage threshold which needs to be exceeded for TIME_HYSTERESIS time when ETPx load drops below one of the threshold to ramp down overload threshold. Value is percentage.* *New value will be provided to ETPA/C during ETPA/C restart and with AoIP related BSC parameter modification.*
TIME_HYSTERESIS **range:** 0 – 30, step 1 **default:** 10	*This is used by ETP to configure Time Hysteresis parameter. This parameter determines time which needs to pass with ETPx load below LOAD_HYSTERESIS to tamp down overload threshold. Value is in seconds.* *New value will be provided to ETPA/C during ETPA/C restart and with AoIP related BSC parameter modification.*

Performance measurements

Performance measurements

Overview

- AoIP introduces completely different transmission mechanism and because of that new measurements along with modifications to existing ones are needed
- 3 new transmission measurements have been implemented:
  - 129 – ETP Ethernet TCSM measurement
  - 130 – AoIP Traffic BSC measurement
  - 131 – AoIP Traffic TCSM measurement
- 3 measurements have been modified and new counters introduced:
  - 4 – Handover measurements (7 new counters)
  - 50 – BSC Level Clear Code Measurement (2 new counters also 11 modified counters introduced)
  - 51 – BSC Level Clear Code PM Measurement (1 new counter)
- 1 measurement from Packet Abis feature can be used also in AoIP
  - 128 – ETP Ethernet BSC measurement

Performance measurements

New measurements and counters 1/15

- 128 – ETP Ethernet BSC measurement:
  - ETP Ethernet BSC measurement – reports Ethernet level statistics provided by ETP
  - This measurement is defined for Packet Abis feature but it is possible to gather measurements from A interface. In case of AoIP only measurements from ETPA units are taken into consideration

Counter	Description
RECEIVED_ETH_FRAMES_ ETP_BSC (128000)	Number of received Ethernet frames (contain errored and non-errored frames). **Unit:** 1000 Ethernet frames. **Updated:** When an Ethernet frame is received. **Use case**: AOIP_DL_FRAME_ERROR_RATE_BSC = ERRORED_ETH_FRAMES_ETP_BSC * 100 / RECEIVED_ETH_ FRAMES_ETP_BSC This KPI shows % of frames with errors in case of AoIP with TC in MGW configuration. It can be used to monitor quality of backhaul network. AOIP_AVG_DL_FRAME_SIZE_BSC = RECEIVED_ETH_ OCTETS_ETP_BSC / ((RECEIVED_ETH_ FRAMES_ETP_BSC - ERRORED_ETH_FRAMES_ETP_BSC) * 1000) Unit of the KPI is Bytes This KPI can be used to estimate efficiency (in terms of user data transmission) of backhaul network usage in case of AoIP with TC in MGW. If this KPI is close to max MTU more user data is transmitted (less BW is used to transmit headers). **In AoIP case counter is aggregated only from ETPA**

Performance measurements

New measurements and counters 2/15

Counter	Description
ERRORED_ETH_FRAMES_ETP_BSC (128001)	Number of received error Ethernet frames. **Unit:** 1000 Ethernet frames. **Updated:** when error Ethernet frame is received. Frame Check Sequence errors, undersized frames, oversized frames, header error and other Ethernet framing errors (errors preventing from being delivered to a higher-layer protocol) are included **Use case**: AOIP_DL_FRAME_ERROR_RATE_BSC AOIP_AVG_DL_FRAME_SIZE_BSC **In AoIP case counter is aggregated only from ETPA**
TRANSM_ETH_FRAMES_ETP_BSC (128002)	Number of transmitted Ethernet frames. **Unit:** 1000 Ethernet frames. **Updated:** When an Ethernet frame is sent . **Use case**: AOIP_AVG_UL_FRAME_SIZE_BSC = TRANSM_ETH_ OCTETS_ETP_BSC/ (TRANSM_ETH_ FRAMES_ETP_BSC * 1000) Unit of the KPI is Bytes This KPI can be used to estimate efficiency (in terms of user data transmission) of backhaul network usage in case of AoIP with TC in MGW. If this KPI is close to max MTU more user data is transmitted (less BW is used to transmit headers). **In AoIP case counter is aggregated only from ETPA**

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Performance measurements

New measurements and counters 3/15

Counter	Description
RECEIVED_ETH_OCTETS_ETP_BSC (128003)	Number of octets in valid frames received on the interface. **Unit:** 1.000.000 octets **Updated:** when an Ethernet frame is received. Use case: AOIP_AVG_DL_GROSS_BIT_RATE_BSC = RECEIVED_ETH_OCTETS_ETP_BSC * 8 / GRANURALITY (in seconds) In case GRANURALITY = 1h: AOIP_AVG_DL_GROSS_BIT_RATE_BSC = RECEIVED_ETH_OCTETS_ETP_BSC * 8 / 3600 Unit of the KPI is Mbps This KPI can be used to monitor if traffic load in DL direction is equally distributed between active ETPs (on ETP object level) or DL gross rate on the A interface (load on all active ETPs) in case of AoIP with TC in MGW. AOIP_AVG_DL_FRAME_SIZE_BSC **In AoIP case counter is aggregated only from ETPA**

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Performance measurements

New measurements and counters 4/15

Counter	Description
TRANSM_ETH_OCTETS_ETP_BSC (128004)	Number of octets in valid frames sent. **Unit:** 1.000.000 octets **Updated:** when an Ethernet frame is sent. **Use case**: AOIP_AVG_UL_GROSS_BIT_RATE_BSC = TRANSM_ETH_OCTETS_ETP_BSC * 8 / GRANURALITY (in seconds) In case GRANURALITY = 1h: AOIP_AVG_UL_GROSS_BIT_RATE_BSC = TRANSM_ETH_OCTETS_ ETP_BSC * 8 / 3600 Unit of the KPI is Mbps This KPI can be used to monitor if traffic load in UL direction is equally distributed between active ETPs (on ETP object level) or to observe UL gross rate on the A interface (load on all active ETPs) in case of AoIP with TC in MGW. AOIP_AVG_UL_FRAME_SIZE_BSC **In AoIP case counter is aggregated only from ETPA**

Performance measurements

New measurements and counters 5/15

- 129 – ETP Ethernet TCSM measurement:
  - New transmission measurement ETP Ethernet TCSM – reports PM data about the availability and quality of Ethernet interface in ETP unit under TCSM supervision
  - The data is reported in Ethernet interface level

Counter	Description
RECEIVED_ETH_FRAMES _ETP_TCSM (129000)	Number of received Ethernet frames (contain errored and non-errored frames). **Unit:** 1000 Ethernet frames. **Trigger event:** Updated when an Ethernet frame is received. **Use case:** AOIP_DL_FRAME_ERROR_RATE_TCSM = ERRORED_ETH_FRAMES_ ETP_TCSM * 100 / RECEIVED_ETH_FRAMES_ETP_TCSM This KPI shows % of frames with errors in case of AoIP with TC in BSS configuration. It can be used to monitor quality of backhaul network. AOIP_AVG_DL_FRAME_SIZE_TCSM = RECEIVED_ETH_ OCTETS_ETP_TCSM/ ((RECEIVED_ETH_ FRAMES_ETP_TCSM - ERRORED_ETH_FRAMES_ETP_TCSM) * 1000) Unit of the KPI is Bytes This KPI can be used to estimate efficiency (in terms of user data transmission) of backhaul network usage in case of AoIP with TC in BSS. If this KPI is close to max MTU more user data is transmitted (less BW is used to transmit headers).

Performance measurements

New measurements and counters 6/15

Counter	Description
ERRORED_ETH_FRAMES_ETP _TCSM (129001)	Number of received error Ethernet frames. **Unit:** 1000 Ethernet frames. **Trigger event:** Updated when error Ethernet frame is received. **Use case:** AOIP_DL_FRAME_ERROR_RATE_TCSM
TRANSM_ETH_FRAMES_ETP_T CSM (129002)	Number of transmitted Ethernet frames. **Unit:** 1000 Ethernet frames. **Trigger event:** Updated when an Ethernet frame is sent. **Use case:** AOIP_AVG_UL_FRAME_SIZE_TCSM = TRANSM_ETH_OCTETS _ETP_TCSM / (TRANSM_ETH_ FRAMES_ETP_TCSM * 1000) Unit of the KPI is Bytes This KPI can be used to estimate efficiency (in terms of user data transmission) of backhaul network usage in case of AoIP with TC in BSS. If this KPI is close to max MTU more user data is transmitted (less BW is used to transmit headers).

Performance measurements

New measurements and counters 7/15

Counter	Description
RECEIVED_ETH_OCTETS_ETP_TCSM (129003)	Number of octets in valid frames received on the interface. **Unit:** 1.000.000 octets **Trigger event:** Updated when an Ethernet frame is received. **Use case:** AOIP_AVG_DL_GROSS_BIT_RATE_TCSM = RECEIVED_ETH_OCTETS_ETP_TCSM * 8 / GRANURALITY (in seconds) In case GRANURALITY = 1h: AOIP_AVG_DL_GROSS_BIT_RATE_TCSM = RECEIVED_ETH_OCTETS_ ETP_TCSM * 8 / 3600 Unit of the KPI is Mbps This KPI can be used to monitor if traffic load in DL direction is equally distributed between active ETPs (on ETP object level) or to observe DL gross rate on the A interface (load on all active ETPs) in case of AoIP with TC in BSS. AOIP_AVG_DL_FRAME_SIZE_TCSM

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Performance measurements

New measurements and counters 8/15

Counter	Description
TRANSM_ETH_OCTETS_ETP_TCSM (129004)	Number of octets in valid frames sent on the interface. **Unit:** 1.000.000 octets **Trigger event:** Updated when an Ethernet frame is sent. **Use case:** AOIP_AVG_UL_GROSS_BIT_RATE_TCSM = TRANSM_ETH_FRAMES_ETP_TCSM * 8 / GRANURALITY (in seconds) In case GRANURALITY = 1h: AOIP_AVG_UL_GROSS_BIT_RATE_TCSM = TRANSM_ETH_OCTETS_ ETP_TCSM * 8 / 3600 Unit of the KPI is Mbps This KPI can be used to monitor if traffic load in UL direction is equally distributed between active ETPs (on ETP object level) or to observe UL gross rate on the A interface (load on all active ETPs)in case of AoIP with TC in BSS. AOIP_AVG_UL_FRAME_SIZE_TCSM

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Performance measurements

New measurements and counters 9/15

- 130 – AoIP Traffic BSC measurement:
  - New transmission measurement AoIP Traffic BSC measurement – reports interface statistics provided by ETP unit controlled by BSC
  - Typically this measurement collects counters from ETPA units

Counter	Description
RTP_PACKETS_RECEIVED_ETP_BSC (130000)	Number of DL RTP packets received successfully from the MGW. **Unit:** packets **Trigger event:** Updated when DL RTP packets are received from MGW. **Use case:** AOIP_DL_PACKET_LOSS_RATE_RTP_BSC = RTP_PACKETS_LOST_ETP_BSC * 100 / (RTP_PACKETS_ RECEIVED_ETP_BSC + RTP_PACKETS_LOST_ETP_BSC) This KPI shows % share of DL RTP packet loss rate in case of AoIP with TC in MGW. It can be used to monitor quality of served calls.
RTP_PACKETS_LOST_ETP_BSC (130001)	The number of lost DL RTP packets that are not received according to received sequence numbers. **Unit:** packets **Trigger event:** After RTP packet validation, the number of lost RTP data packets in DL are calculated and counter value is updated. If the gap in the RTP sequence numbers is < threshold value (100), then the counter is incremented by the amount of missing RTP packets. If the gap in the RTP sequence numbers is => to the threshold value (100), then it is interpreted that there is a jump in the RTP sequence numbers and the counter is not updated. **Use case:** AOIP_DL_PACKET_LOSS_RATE_RTP_BSC

Performance measurements

New measurements and counters 10/15

Counter	Description
RTP_PACKETS_SENT_ETP_ BSC (130002)	Number or UL RTP packets sent to MGW. **Unit:** packets **Trigger event:** Updated when RTP packet is sent to MGW.
RTP_OCTETS_RECEIVED_ETP _BSC (130003)	The total number of received payload octets transmitted in DL RTP packets. **Unit:** octets **Trigger event:** Updated when RTP packet is received successfully from MGW, the counter is incremented by the total number of received payload octets transmitted in RTP data packets **Use case:** AOIP_AVG_DL_USER_BIT_RATE_ETP_BSC = RTP_OCTETS_RECEIVED_ETP_BSC * 8 / GRANURALITY (in seconds) In case GRANURALITY = 1h: AOIP_AVG_DL_USER_BIT_RATE_ETP_BSC = RTP_OCTETS_RECEIVED _ETP_BSC * 8 / 3600 Unit of the KPI is bps This KPI can be used to monitor DL user bit rate load distribution on active ETPs (on ETP object level) or DL user bit rate on the A interface (load on all active ETPs) in case of AoIP with TC in MGW.

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Performance measurements

New measurements and counters 11/15

Counter	Description
RTP_OCTETS_SENT_ETP_BSC (130004)	The total number of sent payload octets transmitted in UL RTP packets. **Unit:** octets **Trigger event:** Updated when RTP packet is sent successfully to MGW, increment the counter by the total number of payload octets transmitted in RTP data packets. **Use case:** AOIP_AVG_UL_USER_BIT_RATE_ETP_BSC = RTP_OCTETS _SENT_ETP_BSC * 8 / GRANURALITY (in seconds) In case GRANURALITY = 1h: AOIP_AVG_UL_USER_BIT_RATE_ETP_BSC = RTP_OCTETS_SENT_ETP _BSC * 8 / 3600 Unit of the KPI is bps This KPI can be used to monitor UL user bit rate load distribution on active ETPs (on ETP object level) or UL user bit rate on the A interface (load on all active ETPs) in case of AoIP with TC in MGW.

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Performance measurements

New measurements and counters 12/15

- 131 – AoIP Traffic TCSM measurement:
  - New transmission measurement AoIP Traffic TCSM measurement – reports A- and Ater interface statistics provided by ETP unit controlled by TCSM.
  - Typically this measurement collects counters from ETPC units

Counter	Description
RTP_PACKETS_RECEIVED_ETP_TCSM (131000)	Number of DL RTP packets received successfully from the MGW. **Unit:** packets **Trigger event:** Updated when DL RTP packets are received from MGW **Use case:** AOIP_DL_PACKET_LOSS_RATE_RTP_TCSM = RTP_PACKETS_ LOST_ETP_TCSM * 100 /(RTP_PACKETS_ RECEIVED_ETP_TCSM + RTP_PACKETS_ LOST_ETP_TCSM) This KPI shows % share of DL RTP packet loss rate in case of AoIP with TC in BSS. It can be used to monitor quality of served calls.
RTP_PACKETS_LOST_ETP_TCSM (131001)	The number of lost DL RTP packets that are not received according to received sequence numbers. **Unit:** packets **Trigger event:** After RTP packet validation, the number of lost RTP data packets in DL are calculated and counter value is updated. When the next sequence number is missing from RTP packet. If the gap in the RTP sequence numbers is < threshold value (100), then the counter is incremented by the amount of missing RTP packets.If the gap in the RTP sequence numbers => threshold value (100), then it is interpreted that there is a jump in the RTP sequence numbers and in this case the counter is not updated. **Use case:** AOIP_DL_PACKET_LOSS_RATE_RTP_TCSM

Performance measurements

New measurements and counters 13/15

Counter	Description
RTP_PACKETS_SENT_ ETP_TCSM (131002)	Number or UL RTP packets sent to MGW. **Unit:** packets **Trigger event:** Updated when RTP packet is sent to MGW.
RTP_OCTETS_RECEIVED _ETP_TCSM (131003)	The total number of received payload octets transmitted in DL RTP packets. **Unit:** octets **Trigger event:** Updated when RTP packet is received successfully from MGW, the counter is incremented by the total number of received payload octets transmitted in RTP data packets This counter can be used to monitor if ETP load sharing is equally distributed. **Use case:** AOIP_AVG_DL_USER_BIT_RATE_ETP_TCSM = RTP_OCTETS_RECEIVED _ETP_TCSM * 8 / GRANURALITY (in seconds) In case GRANURALITY = 1h: AOIP_AVG_DL_USER_BIT_RATE_ETP_TCSM = RTP_OCTETS_RECEIVED _ETP_TCSM * 8 / 3600 Unit of the KPI is bps This KPI can be used to monitor DL user bit rate load distribution on active ETPs (on ETP object level) or to observe DL user bit rate on the A interface (load on all active ETPs) in case of AoIP with TC in BSS.

Performance measurements

New measurements and counters 14/15

Counter	Description
RTP_OCTETS_SENT_ETP_TCSM (131004)	The total number of sent payload octets transmitted in UL RTP packets. **Unit:** octets **Trigger event:** Updated when RTP packet is sent successfully to MGW, the counter is incremented by the total number of sent payload octets transmitted in RTP data packets **Use case:** AOIP_AVG_UL_USER_BIT_RATE_ETP_TCSM = RTP_OCTETS_SENT_ETP_TCSM * 8 / GRANURALITY (in seconds) In case GRANURALITY = 1h: AOIP_AVG_UL_USER_BIT_RATE_ETP_TCSM = RTP_OCTETS_SENT_ETP _TCSM * 8 / 3600 Unit of the KPI is bps This KPI can be used to monitor UL user bit rate load distribution on active ETPs (on ETP object level) or to observe UL user bit rate on the A interface (load on all active ETPs) in case of AoIP with TC in BSS.

Performance measurements

New measurements and counters 15/15

Counter	Description
RTP_PACKETS_TOO_LATE_ETP_TCSM (131005)	Number of DL RTP packets received too late according to timestamp of RTP packet. **Unit:** packets **Trigger event:** Updated when RTP packet is arrived but processing window of the packet is elapsed. Updated when RTP packet is arrived but there is no space left in the de-jitter buffer for processing the packet. Not updated in BSC when Packet Abis over Ethernet/TDM is used with AoIP and transcoding located in MGW. **Use case:** AOIP_DL_PACKET_LOSS_RATE_DUE_TO_DEJITTER_TCSM = (RTP_ PACKETS_TOO_ LATE_ETP_TCSM + RTP_PACKETS_TOO_ EARLY_ETP_TCSM) * 100 / RTP_PACKETS_LOST_ETP_TCSM This KPI gives the % ratio of packet that are lost due to dejitter buffer size to all lost packets. It can be used to optimally set up parameters related to dejitter buffer in case of AoIP with TC in BSS.
RTP_PACKETS_TOO_EARLY_ETP_TCSM (131006)	Number of DL RTP packets received too early according to timestamp of RTP packet. **Unit:** packets **Trigger event:** Updated when RTP packet is arrived but there is no space left in the de-jitter buffer for processing the packet. Updated when RTP packet is arrived but processing window of the packet is elapsed Not updated in BSC when Packet Abis over Ethernet/TDM is used with AoIP and transcoding located in MGW. **Use case:** AOIP_DL_PACKET_LOSS_RATE_DUE_TO_DEJITTER_TCSM

Performance measurements

Modified measurements with new counters 1/6

- 4 – Handover measurement:

Counter	Description
BSS_INT_HO_REQ_ REQUESTS (004244)	Number of *BSS Internal Handover Required requests* when IP based user plane is supported in the A interface. *BSS Internal Handover Required request* message is new BSSAP SCCP message introduced by AoIP feature. This message is sent from the BSS to the MSC to indicate that for a given MS, which already has dedicated radio resource(s) assigned, a handover is required to a cell in the same BSS, for the reason given by the cause element. This message is used when the intra-BSS handover implies a Codec Type or a Codec Configuration or an A-interface type in the target cell, which is incompatible with the one used in the source cell. The source cell and target cell can be the same cell. **Trigger event:** Updated when BSSMAP Internal Handover Required message is sent to MSS from BSC **Use case:** BSS_INT_HO_SUCCESS_RATE = BSS_INT_HO _SUCCESSFUL * 100 / BSS_INT_HO_REQ_REQUESTS This KPI can be used to monitor new internal handovers success ratio [%]. BSS_INT_HO_REJ_RATE = BSS_INT_HO_REQ_REJECTIONS * 100 / BSS_INT_HO_REQ_REQUESTS This KPI can be used to monitor codec and/or interfaces matching in call serving cells [%]. If rejection ratio is high there should be considered change of the supported codec list options.

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Performance measurements

Modified measurements with new counters 2/6

Counter	Description
BSS_INT_HO_ COMMANDS (004245)	Number of *BSS Internal Handover Commands* received from the MSC when IP based user plane is supported in the A interface. *BSS Internal Handover Command* message is new BSSAP SCCP message introduced by AoIP feature and can be send if and only if both the BSS and the MSC support an IP based user plane interface. This message is sent from the MSC to the BSS and contains the “Speech Codec (MSC Chosen)” indicating the Codec(s) chosen by the MSC and the “AoIP Transport Layer Access (MGW)” as a response to *BSS Internal Handover Required requests* **Trigger event:** Updated when BSSMAP Internal Handover Command message is received on BSC from MSS **Use case:** BSS_INT_HO_FAILURE_RATE = BSS_INT_HO _FAILURES * 100 / (BSS_INT_HO_COMMANDS + BSS_INT_HO_ENQUIRY_REJ) This KPI can be used to monitor Internal handover failure rate [%].
BSS_INT_HO_ FAILURES (004246)	Number of BSS Internal Handover failures sent by the BSC when IP based user plane is supported in the A interface. (Count of BSSAP Handover Failure messages related to the Internal Handover execution procedure) **Trigger event:** Updated when BSC sends BSSMAP Handover Failure message to MSC **Use case:** BSS_INT_HO_FAILURE_RATE

Performance measurements

Modified measurements with new counters 3/6

Counter	Description
BSS_INT_HO_REQ_REJECTIONS (004247)	Number of *BSS Internal Handover Required Rejections* received from the MSC when IP based user plane is supported in the A interface *BSS Internal Handover Required Rejection* message is new BSSAP SCCP message introduced by AoIP feature and can be send if and only if both the BSS and the MSC support an IP based user plane interface. **Trigger event:** Updated when Internal Handover Required Reject is sent from MSS to BSC **Use case:** BSS_INT_HO_REJ_RATE
BSS_INT_HO_SUCCESSFUL (004248)	Number of successful BSS Internal handover procedures when IP based user plane is supported in the A interface. (Count of *Handover Complete* messages related to the internal handover execution procedure) **Trigger event:** Updated when BSC sends BSSMAP Handover Complete message to MSS **Use case:** BSS_INT_HO_SUCCESS_RATE

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Performance measurements

Modified measurements with new counters 4/6

Counter	Description
BSS_INT_HO_ENQUIRY (004249)	Number of *BSS Internal Handover Enquiry* messages received from the MSC when IP based user plane is supported in the A interface. BSS Internal Handover Enquiry message is sent by MSS to distant BSS to inform that Internal HO is needed to matching codec with the local side of the connection to ensure that TrFO/TFO can be in use. **Trigger event:** Updated when BSC receives BSSMAP Internal Handover Enquiry message from MSS **Use case:** BSS_INT_HO_MSS_REQ_FAILURE_RATE = BSS_INT_HO_ENQUIRY_REJ * 100 / BSS_INT_HO_ENQUIRY This KPI can be used to monitor BSC internal handover failure rate in case of codec change to non matching configuration and MSS forced HO on distant side of the connection (required to allow TFO/TrFO) [%].
BSS_INT_HO_ENQUIRY_REJ (004250)	Number of *BSS Internal Handover Enquiry rejections* when IP based user plane is supported in the A interface (Count of *Handover Failure* messages related to the *Internal Handover Enquiry*) **Trigger event:** Updated when BSC sends BSSMAP Internal Handover Enquiry Reject message to MSS **Use case:** BSS_INT_HO_MSS_REQ_FAILURE_RATE

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Performance measurements

Modified measurements with new counters 5/6

- 50 – BSC Level Clear Code Measurement:

Counter	Description
NBR_OF_CALLS (0500409)	Number of Assignment Failure or Handover Failure messages that indicates unavailability of requested codec type (or codec configuration). **Trigger event:** Updated when BSC sends Handover Failure or Assignment Failure message to MSC with BSSMAP cause set as "Requested Codec Type or Codec Configuration Unavailable“.
NBR_OF_CALLS (0500945)	Number of Internal Handover Required Reject messages from MSC which report that requested redundancy level is not supported. **Trigger event:** Updated when BSC receives Internal Handover Required Reject from MSC with BSSMAP cause set as "Requested Redundancy Level Not Supported“.

Performance measurements

Modified measurements with new counters 6/6

- 51 – BSC Level Clear Code PM Measurement:

Counter	Description
NBR_OF_CALLS (051220)	Number of Internal Handover Required messages which report response to an MSS internal handover enquiry message. **Trigger event:** Updated when BSC sends BSSMAP Internal Handover Required message to MSS in response to Internal Handover Enquiry message received from MSS. BSSMAP cause is set as "Response to an INTERNAL HANDOVER ENQUIRY message"

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Performance measurements

Modified counters 1/4

- 50 – BSC Level Clear Code Measurement:

Counter	Description
NBR_OF_CALLS (0500630)	Number of Assignment Failure or Handover Failure messages which report that requested A-interface type is not supported. **Trigger event:** Updated when BSC sends Handover Failure or Assignment Failure message to MSC with BSSMAP cause set as "Requested A-Interface Type Not Supported“
NBR_OF_CALLS (0500631)	Number of Assignment Failure or Handover Failure messages which report that requested redundancy level is not supported. **Trigger event:** Updated when BSC sends Handover Failure or Assignment Failure message to MSC with BSSMAP cause set as "Requested Redundancy Level Not Supported“
NBR_OF_CALLS (0500632)	Number of Assignment Failure or Handover Failure messages which report that requested codec type or codec configuration is not supported. **Trigger event:** Updated when BSC sends Handover Failure or Assignment Failure message to MSC with BSSMAP cause set as "Requested Codec Type or Codec Configuration Not Supported“
NBR_OF_CALLS (0500633)	Number of Assignment Failure or Handover Failure messages which report that requested A-interface type is unavailable. **Trigger event:** Updated when BSC sends Handover Failure or Assignment Failure message to MSC with BSSMAP cause set as "Requested A-Interface Type Unavailable“

Performance measurements

Modified counters 2/4

- 50 – BSC Level Clear Code Measurement:

Counter	Description
NBR_OF_CALLS (0500634)	Number of Internal Handover failures due to T25 timer expiry. **Trigger event:** Updated when T25 timer expires on BSC waiting for Internal Handover Command from MSC. This is incremented only when no more Handover re-tries are allowed from the BSC or the number of re-tries have exceed at the prescribed limit.
NBR_OF_CALLS (0500717)	Number of Assignment Failure or Handover Failure messages which report that call identifier is already allocated. **Trigger event:** Updated when BSC sends Handover Failure or Assignment Failure message to MSC with BSSMAP cause set as "Call Identifier Already Allocated“
NBR_OF_CALLS (0500718)	Number of BSSAP Handover Failure messages which report that requested terrestrial resource is unavailable. **Trigger event:** Updated when BSC sends Handover Failure message to MSC with BSSMAP cause set as "Requested Terrestrial Resource Unavailable"

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Performance measurements

Modified counters 3/4

- 50 – BSC Level Clear Code Measurement:

Counter	Description
NBR_OF_CALLS (0500941)	Number of Internal Handover Required Reject messages from MSC which report that requested codec type or codec configuration is not supported. **Trigger event:** Updated when BSC receives Internal Handover Required Reject from MSC with BSSMAP cause set as "Requested Codec Type or Codec Configuration Not Supported“.
NBR_OF_CALLS (0500942)	Number of Internal Handover Required Reject messages from MSC which report that requested A-interface type is not supported. **Trigger event:** Updated when BSC receives Internal Handover Required Reject from MSC with BSSMAP cause set as "Requested A-Interface Type Not Supported“.
NBR_OF_CALLS (0500943)	Number of Internal Handover Required Reject messages from MSC which report that requested A-interface type is unavailable. **Trigger event:** Updated when BSC receives Internal Handover Required Reject from MSC with BSSMAP cause set as "Requested A-Interface Type Unavailable“.

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Performance measurements

Modified counters 4/4

- 50 – BSC Level Clear Code Measurement:

Counter	Description
NBR_OF_CALLS (0500944)	Number of Internal Handover Required Reject messages from MSC which reported that requested codec type or codec configuration is unavailable. **Trigger event:** Updated when BSC receives Internal Handover Required Reject from MSC with BSSMAP cause set as "Requested Codec Type or Codec Configuration Unavailable“.

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Feature impact analysis

Feature impact analysis

Feature impact analysis 1/2

Feature impact	How to measure
Replacement of TDM transport with IP/Ethernet backhaul may affect the existing quality: thus basic (and existing) KPIs should be observed to proof that the AoIP does not degrade the network quality	**KPI:** TCH Traffic in Erlang (trf_24c) TCH traffic share of HR AMR calls (trf_122) Dropped Call Rate (dcr_3i) Call Drop cause distribution (based on Clear Codes) Call Setup Success Rate (cssr_2) TCH blocking (blck_8h) Handover Failure Rate (hfr_1) Intra-cell TCH-TCH HO success % (hsr_9) HO attempts (ho_13a) HO causes distribution Bad DL Qual (x=5,6,7) (qdl_2) Bad UL Qual (x=5,6,7) (qul_2) Bad UL FER Rate (>4,2%) (ulq_3)

Feature impact analysis

Feature impact analysis 2/2

Feature impact	How to measure
Bandwidth usage on A interface over IP needs to be monitored	**KPI:** **In case of AoIP with TC in MGW** DL_GROSS_BIT_RATE_BSC UL_GROSS_BIT_RATE_BSC **In case of AoIP with TC in BSS** DL_GROSS_BIT_RATE_TCSM UL_GROSS_BIT_RATE_TCSM
PDV buffering must be compromised between packet loss ratio due PDV and increased delay. New KPIs are defined to help in that process	**KPI:** DL_RTP_PACKET_LOSS_TCSM_PDV
New types of Internal Handovers are introduced with AoIP feature. To monitor these HO new KPIs are defined.	**KPI:** BSS_INT_HO_SUCCESS_RATE BSS_INT_HO_REJ_RATE BSS_INT_HO_FAILURE_RATE BSS_INT_HO_MSS_REQ_FAILURE_RATE

Thank you for your attention!

Nokia Siemens Networks

Office Address:
NOKIA SIEMENS NETWORKS
NWS LTE RA E2E Mgmt SA NE
Pl. Bema 2, 1st floor
PL 50-265 Wroclaw

Phone: +48 728 974 487
e-mail: piotr.malatynski@nsn.com

Piotr Malatynski
NWS LTE RA E2E Mgmt SA NE GSM & LTE Migration