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TECHNICAL REPORT

Universal Mobile Telecommunications System (UMTS); Bandwidth And Resource Savings (BARS) and speech enhancements for Circuit Switched (CS) networks (3GPP TR 23.977 version 16.0.0 Release 16)

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Contents

Foreword

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1 Scope

The objective of this technical report is to identify the full set of requirements for bandwidth and resource savings and improved speech quality, with specific consideration to networks supporting A/Gb mode and the bearer independent circuit-switched core network (BICN). The different architectural solutions to meet these requirements will be assessed.

Consideration shall be made to existing architectures and solutions to provide harmony between 2G nodes, UMTS nodes and external networks (PSTN/ISDN). Backward compatibility to existing solutions and ease of network introduction/upgrade shall be given high importance.

2 References

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

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document.*

[1] 3GPP TS 23.002: "Network architecture".

[2] 3GPP TS 23.153: "Out of band transcoder control; Stage 2".

[3] 3GPP TS 23.053: "Tandem Free Operation (TFO); Service description; Stage 2".

[4] 3GPP TS 28.062: "Inband Tandem Free Operation (TFO) of speech codecs; Service description; Stage 3".

[5] 3GPP TS 26.103: "Speech codec list for GSM and UMTS".

[6] 3GPP TR 26.975: "Performance Characterization of the AMR Speech Codec".

[7] 3GPP TR 26.976: "Performance characterization of the Adaptive Multi-Rate Wideband (AMR-WB) speech codec".

[8] 3GPP TS 26.102: "Mandatory speech codec; Adaptive Multi-Rate (AMR) speech codec; Interface to Iu, Uu and Nb".

[9] Void.

[10] 3GPP TS 48.060: "In-band control of remote transcoders and rate adaptors for full rate traffic channels".

[11] 3GPP TS 48.061: "In band control of remote transcoders and rate adaptors for half rate traffic channels".

[12] 3GPP TS 52.021: "Network Management (NM) procedures and messages on the A-bis interface".

[13] 3GPP TS 29.163: "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks".

[14] ITU-T Recommendation Q.1912.5: "BICC SIP Interworking".

[15] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[16] 3GPP TS 29.007: "General requirements on interworking between the Public Land Mobile Network (PLMN) and the Integrated Services Digital Network (ISDN) or Public Switched Telephone Network (PSTN)".

[17] 3GPP TS 48.008: "Mobile Switching Centre - Base Station System (MSC-BSS) interface; Layer 3 specification".

[18] 3GPP TS 26.101: "Mandatory speech codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec frame structure".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the definitions in TR 21.905 [15] apply as well as the following terms and definitions.

Bearer Independent Core Network : This term refers to a core network (CN) comprised of MSC Server, CS-MGW and GMSC Server nodes to support MSC and GMSC functionality, as defined in TS 23.002 [1].

Codec Configuration : The Codec Configuration of a codes type ,like AMR, includes mainly the Active Codec Set, the setting of the OM flag and DTX parameters, etc.

MIPS: Mega (Million) Instructions Per Second. This is a measure for the required DSP capacity. It is here in this context related to the "ETSI-DSP" as defined in 3GPP SA4 for the complexity characterisation of the 3GPP Speech Codec algorithms.

3.2 Symbols

For the purposes of the present document, the following symbols apply:

Ater	the reference point internal to the BSC functional entity, between the transcoders and the rest of the BSC functions.
Abis	interface between the BTS and the BSC.

3.3 Abbreviations

For the purposes of the present document, the abbreviations in TR 21.905 [15] apply as well as the following abbreviations:

BICN	Bearer Independent Core Network
BICC	Bearer Independent Call Control
DSP	Digital Signal Processor
DC	Decoding
MSC-S	MSC Server
OoBTC	Out of Band Transcoder Control
PT	TFO Protocol Termination
R	Reframing
TC	Transcoding
UP	User Plane Termination

4 Network deployment scenarios to be studied

4.1 GSM network architecture before Release 4

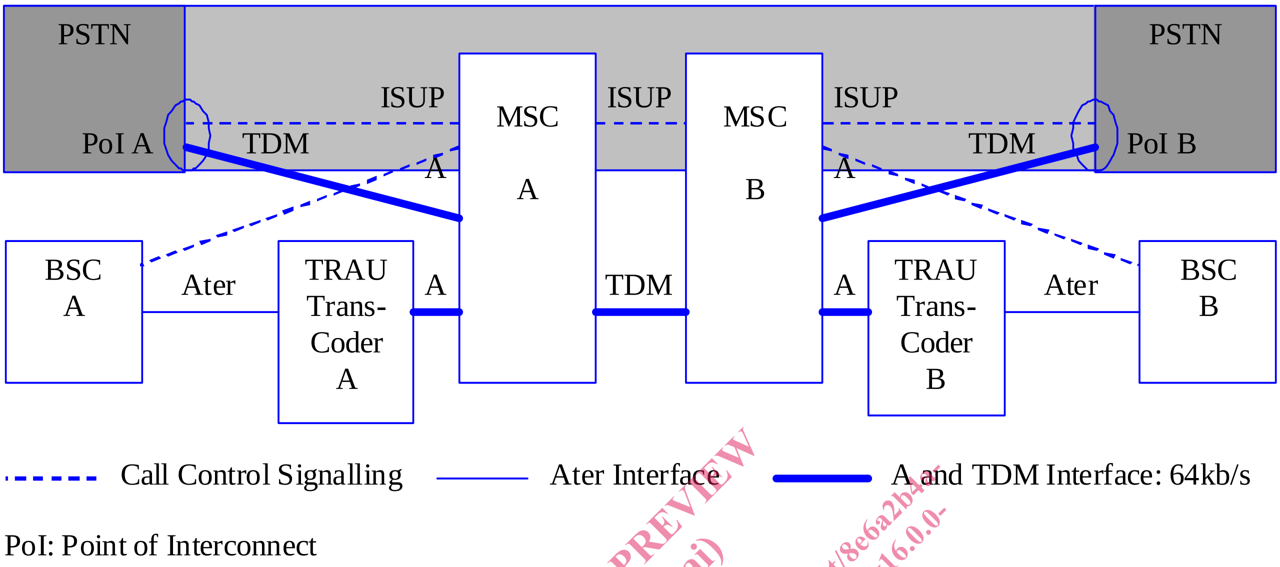


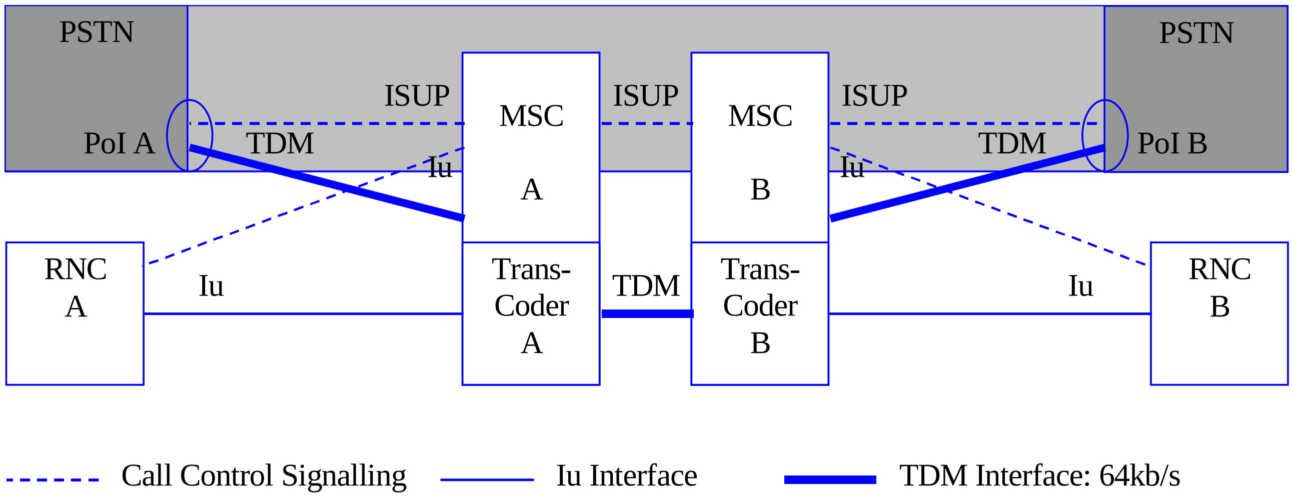
Figure 4.1-1: GSM Network Architecture before release 4

In GSM networks according to releases before release 4 the MSCs are interconnected on the user plane by TDM links (real or virtual) with 64 kb/s for speech traffic. The only speech codec type known between MSCs is G.711 'PCM'. There are typically several Points-of-Interconnect to the underlying PSTN, with 64kb/s for the speech traffic in PCM. The MSCs control and interconnect the BSCs via the A-Interface (user plane and control plane), but they have no direct influence on the Codec Type selected by the BSC on the GSM radio access. The MSC can make a suggestion on the Codec Type, but the BSC decides finally. The MSCs have no means at all to signal the Codec Configuration to the BSCs or between MSCs. This is a drawback.

The transcoders belong logically to the GSM_BSS. Speech is transported on the Ater interface in compressed form using the same codec type and configuration as on the radio interface.

Tandem Free Operation (TFO) is defined on PCM links for all GSM Codec Types. TFO allows by inband signalling to 'tunnel' the compressed speech through the TDM core network. TFO provides the possibility to bypass and omit the encoding functions, saves DSP resources, improves the speech quality in mobile-mobile calls, allows new speech services like wideband speech, but does not provide transmission cost saving.

4.2 UMTS network architecture in Release 99



PoI: Point of Interconnect

Figure 4.2-1: UMTS network architecture in Release 99

In UMTS networks according to release 99 the MSCs are interconnected on the user plane by TDM links (real or virtual) with 64 kb/s for speech traffic. The only speech codec type known between MSCs is G.711 'PCM'. There are typically several Points-of-Interconnect to the underlying PSTN, with 64kb/s for the speech traffic in PCM.

The MSCs control and interconnect the RNCs via the Iu-Interface (user plane and control plane).

The MSC selects and commands the Codec Type on the UTRAN radio access and makes a suggestion on the Codec Configuration, but the RNC can select a sub-configuration.

The Transcoders are located on central places physically and logically 'inside' the mobile core network as integral parts of the MSCs. They are controlled by the MSCs via internal interfaces. But also the RNC controls the transcoder via the Iu interface (Iu_Init). Speech is transported on the Iu-interface in compressed form using the same codec type and configuration as on the radio interface.

The MSCs have no means at all to signal the Codec Configuration between MSCs. This is a drawback.

Tandem Free Operation (TFO) is defined on PCM links for all UMTS Codec Types (there is only UMTS_AMR and UMTS_AMR2). TFO allows by inband signalling to 'tunnel' the compressed speech through the TDM core network. TFO provides the possibility to bypass and omit the encoding functions, saves DSP resources, improves the speech quality in mobile-mobile calls, allows new speech services like wideband speech, but does not provide transmission cost saving. It is possible to have a combined GSM/UMTS core network with MSCs supporting both the Iu interface towards RNCs and the A-interface towards BSCs.

4.3 Packet transport network between MGWs in an A/Iu mode BICN

NOTE: Since we consider only speech telephony services in this TR the Gb interface has no relevance.

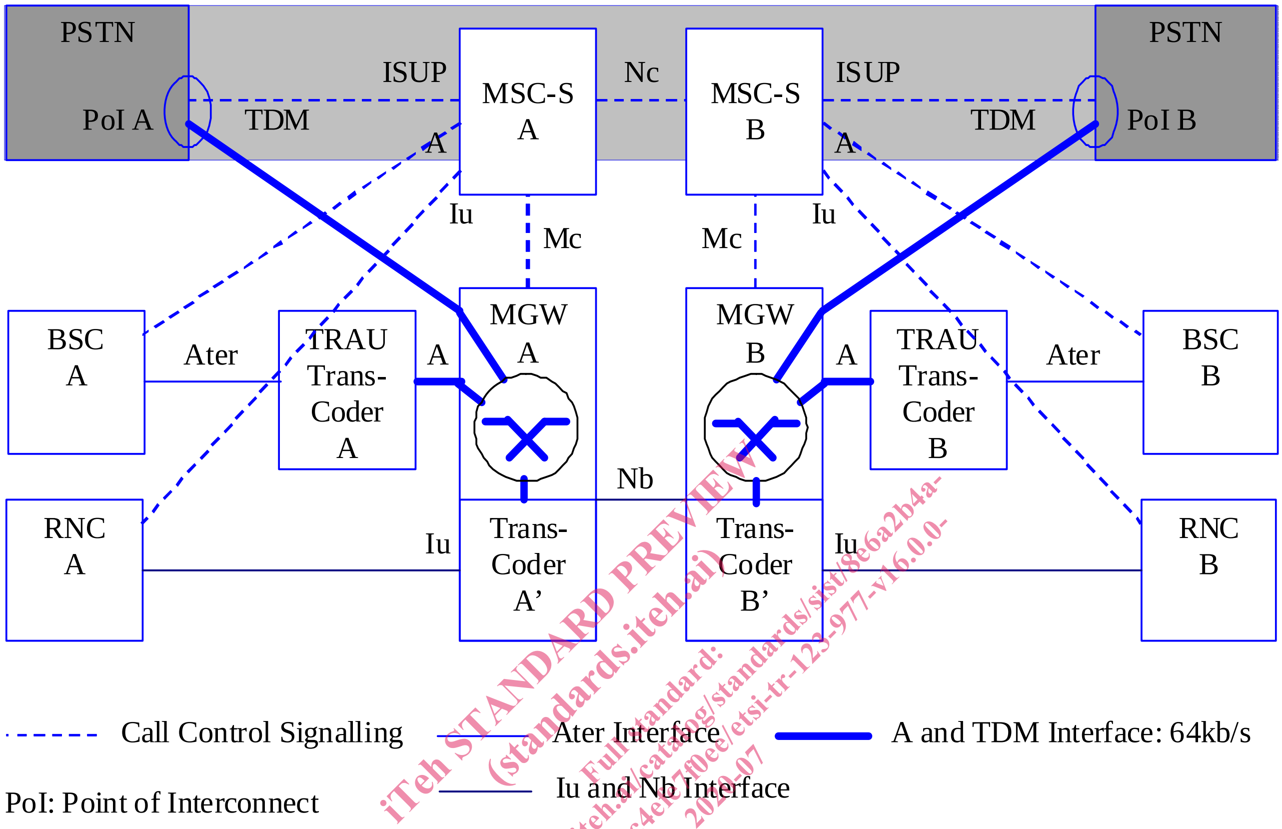


Figure 4.3-1: Bearer Independent Core Network with A- and Iu-interfaces from Release 4 onwards

The mobile Core Network from release 4 onwards has a layered architecture with BICC and OoBTC/TrFO on the Nc/Nb interface or TFO on the Nb interface and provides the means to transport speech in compressed form on the Nb interface.

The MSC-Ss know, negotiate and select the speech Codec Types and Configurations on the Iu interfaces. The MSC-Ss also know, negotiate and select speech Codec Types and Configurations on the Nb Interface.

- This may lead to Transcoder free operation (TrFO) with compressed speech at the Nb interface.

- If the MSC-Ss determine G.711 as the codec used between the MGWs, then the MGWs may afterwards establish TFO at the Nb interface. In this case the transcoders in the MGWs know and negotiate the speech codec configuration on the Nb interface, and they inform the MSC-Server of this configuration indicating that TFO is possible. If the transcoder is in the BSCs, the BSCs know and select the speech codec type and configuration on the A-ter interface to enable TFO operation on the A interface.

The RNC accepts the commanded Codec Type and Configuration.

The MSC-Ss suggest also the speech Codec Type to be used on the Ater interface, but the BSC has the final decision and determines the initial Codec Type and Configuration for the GSM radio interface and the Ater interface. The MSC-Ss cannot communicate the preferred Codec Configuration to the BSCs in a direct way. The MSC-Ss can discover the Codec Type and Configuration from the BSCs via the TFO procedures at the corresponding MGW. The MSC-Ss can then direct interworking procedures between TFO on an A interface or other TDM link and either OoBTC or TFO associated with an Nb interface to optimally allocate the speech transcoder functions.

The MGWs host the transcoding and interworking between compressed speech on Nb or Iu and the legacy 'PCM' with or without TFO on A and TDM interfaces. Points-of-Interconnect to the PSTN are typically provided at every MGW. MGWs may be geographically distributed to minimise the length of the speech path inside the PSTN.

Bandwidth efficient transmission is always provided on the Ater- and on the Iu-interfaces, where the Iu allows a slightly higher efficiency in DTX due to its packet based transport structure (ATM or IP).

The bandwidth efficiency on the Nb-Interface depends on the selected Codec Type. It can be as on Iu (when TrFO is used) or it can be 64kb/s for G.711.In the latter case, the bandwidth efficiency on the Nb-interface is always 64kb/s for PCM, even when a compressed Codec Type has been selected by using TFO. This is a drawback.

OoBTC may lead to a Transcoder free Operation (TrFO) with high bandwidth efficiency on all user planes for UE-to-UE calls. For UE-to-PSTN calls at least the major part of the speech path can be realised in compressed form (TrFO-link, Transcoder at the Edge of the CN).

For any call transiting the Nb interface, both OoBTC and TFO procedures may apply. Harmonization procedures between OoBTC and TFO provide the necessary interworking, achieving the same speech quality benefits provided separately by either TrFO or TFO. OoBTC and TFO for MS-to-UE and MS-to-MS calls that traverse a packet transport network over Nb may lead to a combination of TrFO/TFO and TFO operation on the Nb and A interface / TDM portions of the speech path, respectively, with high bandwidth efficiency on all but the A interface and TDM portions of the speech path, except when TFO is used over Nb interface. OoBTC and TFO for MS-to-PSTN calls that traverse a packet transport network over Nb may also provide for high bandwidth efficiency on any Ater, Iu and Nb portions of the speech path, except when TFO is used over Nb interface.

4.4 TDM transit network between A/Iu mode PLMNs

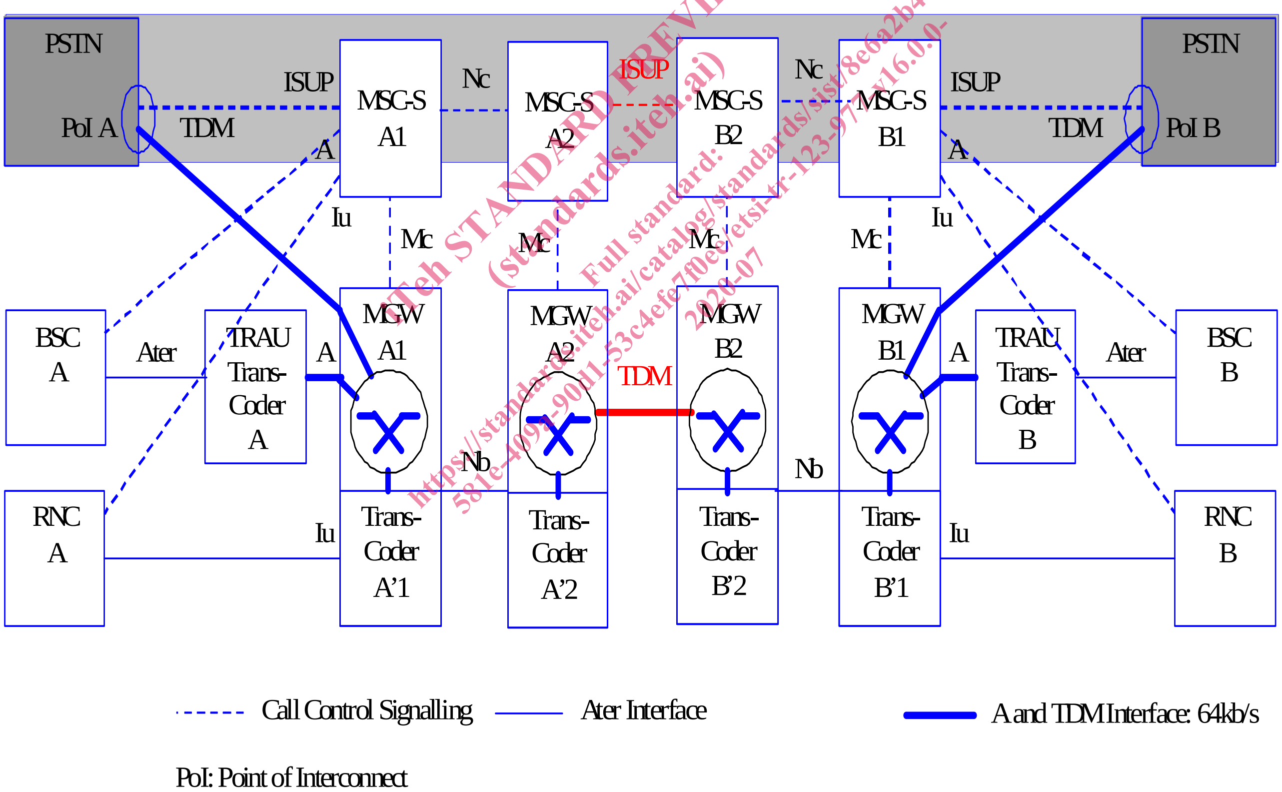


Figure 4.4-1: TDM Transit network between PLMNs from Release 4 onwards

This architecture shows two mobile Core Networks (BICNs) of Release 4 or 5 in layered architecture, with BICC and OoBTC on the Nc interface or TFO on the Nb interface and speech in compressed form on the Nb interface, connected by a legacy ISUP signalling and TDM with 64kb/s for speech (G.711). All features as explained above for one BICN are of course valid inside each BICN and are not further reprinted here in all details.

TFO on the TDM interface between the BICNs (here between MGW A2 and MGW B2) can be used to exchange the compressed speech parameters between both BICNs. By that, end-to-end transcoding free operation is possible in any combination of mobile-to-mobile calls, provided that no In-Path_Equipment prevents the establishment of TFO on these links. Also "Transcoder at the edge" can be provided in any combination of mobile-to-PSTN calls, regardless whether the Point-of-Interconnect to the PSTN is inside the BICN where the mobile is connected, or in the other BICN. Cost efficient transmission is possible within each BICN, but of course not (directly) on the TDM link between the BICNs, except when TFO is used on Nb interface. The resulting speech quality should be identical to the one achievable within one BICN. In all call scenarios the optimal speech quality can be achieved.

Within each BICN, for TrFO, the MSC-Ss know, negotiate and select the speech Codec Types and Configurations on the Nb and Iu interfaces and suggest also the speech Codec Type to be used on the Ater interface.

- This may lead to Transcoder free operation (TrFO) with compressed speech at the Nb interface.
- If the MSC-Ss determine G.711 as the codec used between the MGWs, then the MGWs may afterwards establish TFO at the Nb interface. In this case the transcoders in the MGWs know and negotiate the speech codec configuration on the Nb interface, and they inform the MSC-Server of this configuration indicating that TFO is possible. If the transcoder is in the BSCs, the BSCs know and select the speech codec type and configuration on the A-ter interface to enable TFO operation on the A interface.

For TrFO, the MSC-Ss of one BICN cannot negotiate Speech Codec Type/Configuration directly with the MSC-Ss of the other BICN due to the ISUP connection between them. OoBTC-signalling therefore ends at the border MGWs (here MGW A2 and MGW B2). TFO inband signalling connects both BICNs and provides OoBTC-compatible means to exchange the Codec Lists and to identify the optimal Codec Type and Configuration. In this way a complete end-to-end Codec List negotiation is achieved.

The main difference between OoBTC- and TFO-signalling is, that one is performed before call setup and the other immediately after call setup. As both Core Networks could select different, incompatible Codec Types/Configurations that TFO cannot in all cases establish immediately. The Codec Mismatch situation and the Optimal Codec Type/Configuration gets known to both BICNs by TFO signalling and then it might be required that one or both BICNs perform an In-Call-Modification of the Codec Type/Configuration to achieve end-to-end transcoding free transport.

It may be noted here for completeness that also "inside" the ISUP/TDM connection between the shown BICNs another BICN may be "hidden" with TFO capability to the external world. This hidden BICN could have the same OoBTC and Codec Types/Configurations and by that support high bandwidth efficiency on long trunks without any loss of speech quality.

4.5 Packet transport transit network between PLMNs

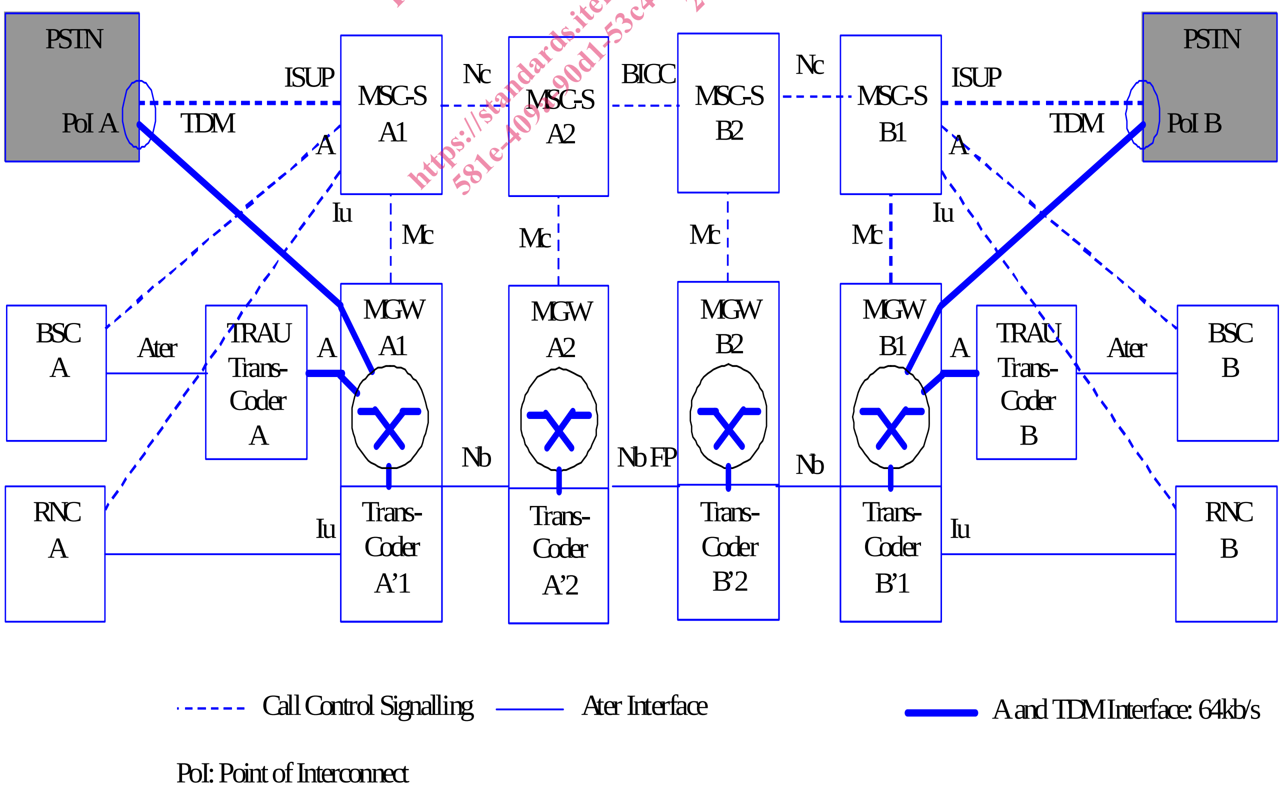


Figure 4.5-1: Packet transport transit network between PLMNs of REL5