Technical Specification Group Services and System Aspects **TSGS#22(03)0713**

Meeting #22, Maui, Hawaii, USA, 15-18 December 2003

Presentation of Specification to TSG or WG

Presentation to:	**TSG SA Meeting #22**
Document for presentation:	**TR 23.977, Bandwidth and Resource Savings and Speech Enhancements for CS Networks (BARS)**
Version	**1.0.0**
Presented for:	**Information**

Abstract of document:

TR 23.977 is a study to identify the requirements for bandwidth and resource savings and improved speech quality, with specific consideration to networks supporting A/Gb mode and the BICN. The different architectural solutions to meet these requirements are assessed.

TR 23.977 is more than 50% complete and is presented to SA for information.

TR 23.977 identifies the network deployment scenarios and call scenarios to be studied. GSM and UMTS network architecture of before Release 4 and of Release 4 and later are studied. The call scenarios cover mobile to mobile, mobile to PSTN as well as roaming call scenarios. All possible combinations of calls between BSCs and/or RNCs are included. In addition requirements and architectural solutions for resource savings, for bandwidth savings, for speech quality improvements as well as for avoiding duplication in transcoder development are described.

Changes since last presentation to SA

This is the first presentation to SA of TR 23.977.

Outstanding Issues:

Further work in the areas of

- Network Deployment Scenarios
- Call Scenarios
- Requirements and Architectural Solutions for Resource Savings
- Requirements and Architectural Solutions for Bandwidth Savings
- Requirements and Architectural Solutions for Speech Quality Improvements
- Requirements and Architectural Solutions for Avoiding Duplication in Transcoder Development

Are required to complete the TR.

Contentious Issues:

None at present.

3GPP TR 23.977 V1.0.0 (2003-12)

Technical Report

**3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Bandwidth and Resource Savings and Speech Enhancements for CS Networks (BARS)
(Release 6)**

GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

Keywords

<TBD>

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Contents

Foreword

This Technical Report has been produced by the 3rd Generation Partnership Project (3GPP).

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the document.

Introduction

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

[1] TS 23.002 Network Architecture

[2] TS 23.153 Out of Band Transcoder Control; Stage 2

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

[4] TS 28.062 Tandem Free Operation (TFO); Service description; Stage 3

[5] TS 26.103 Speech codec list for GSM and UMTS

[6] TR 26.975 Performance characterization of the Adaptive Multi-Rate (AMR) speech codec

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

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

[9] TS 26.103 Speech codec list for GSM and UMTS

3 Definitions, Symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply.

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 [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..

3.2 Symbols

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

[Editors Note: None defined currently]

3.3 Abbreviations

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

BICN Bearer Independent Core Network
MSC-S MSC Server

4 Network deployment scenarios to be studied

[Editors Note: It is proposed that a section of the TR be reserved for documenting typical network deployment scenarios. This will provide background for the discussion. Particular attention should be given to the relative locations of the BSC, Transcoder, MGW and Call Server in the following network deployment scenarios:

GSM current network deployments scenarios

How BICN likely could apply in those network deployment scenarios

Typical network architectures that involve Interworking to the PSTN or other fixed (transit) networks. These networks could be TDM or packet]

4.1 GSM Network Architecture before Release 4

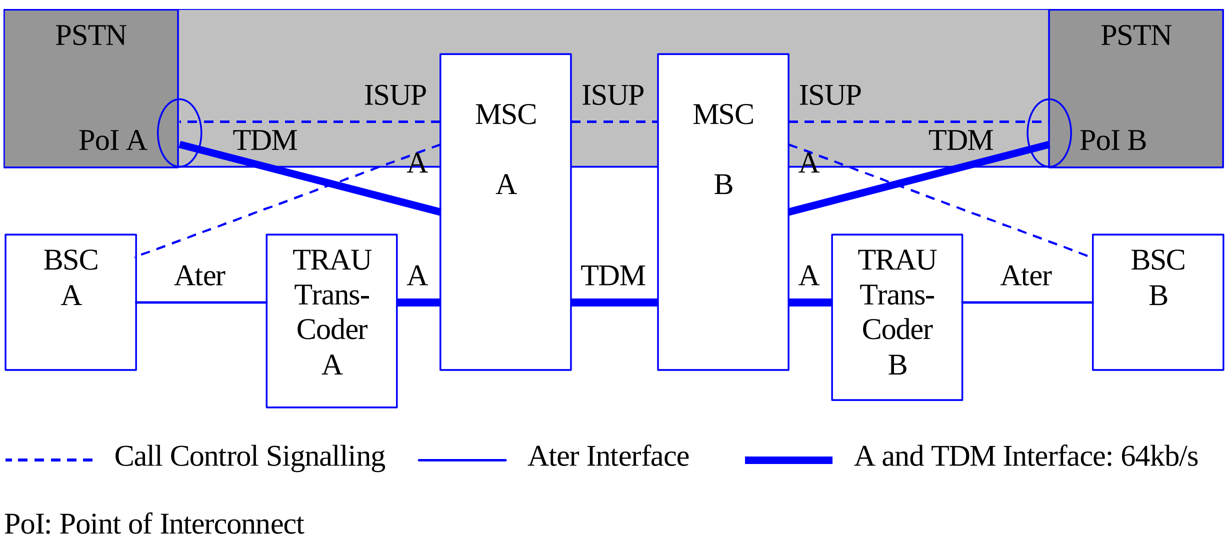


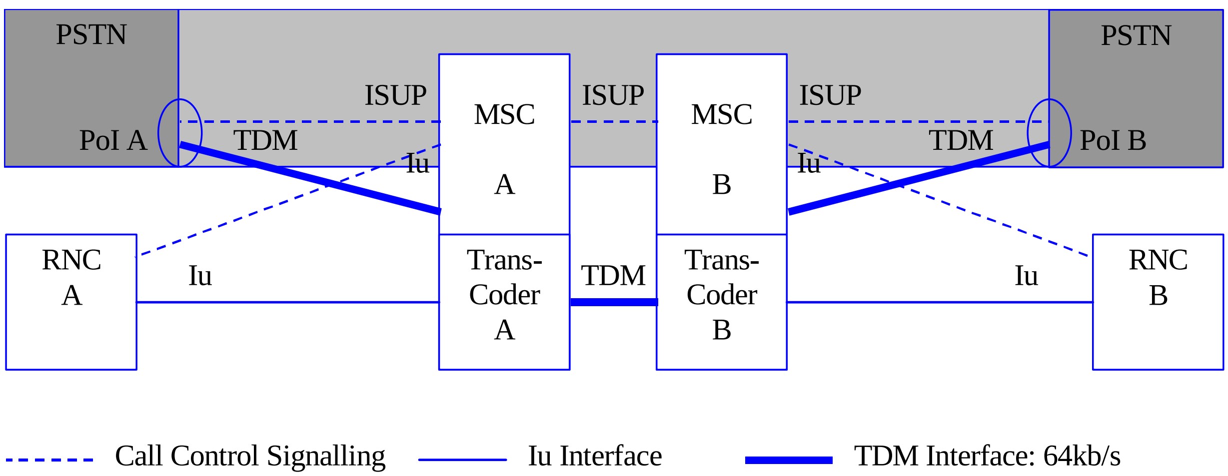
Figure 4.1-1 GSM Network Architecture before release 4

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 releases in 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 transcode 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.

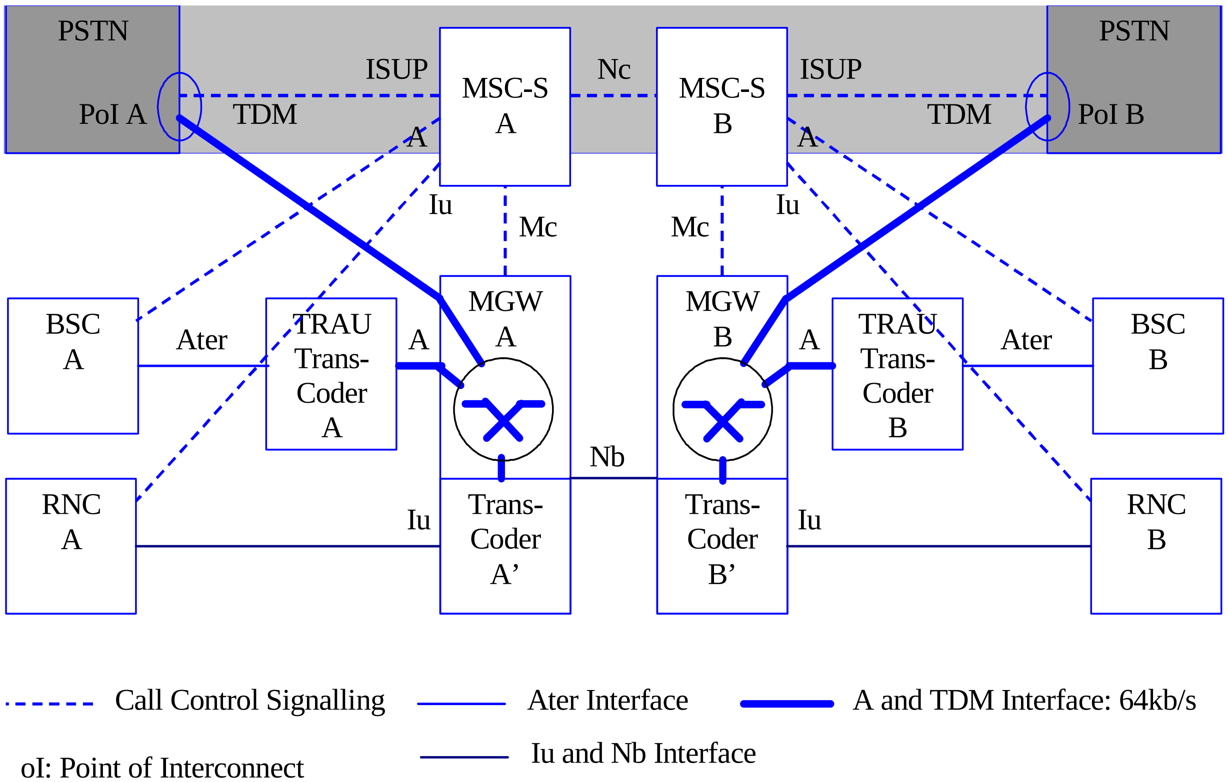


Figure 4.3-1 Bearer Independent Core Network with A- and Iu-Interfaces from Release 4 onwards

[Editors Note: Need to update this picture to show the BICN as in next section]

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

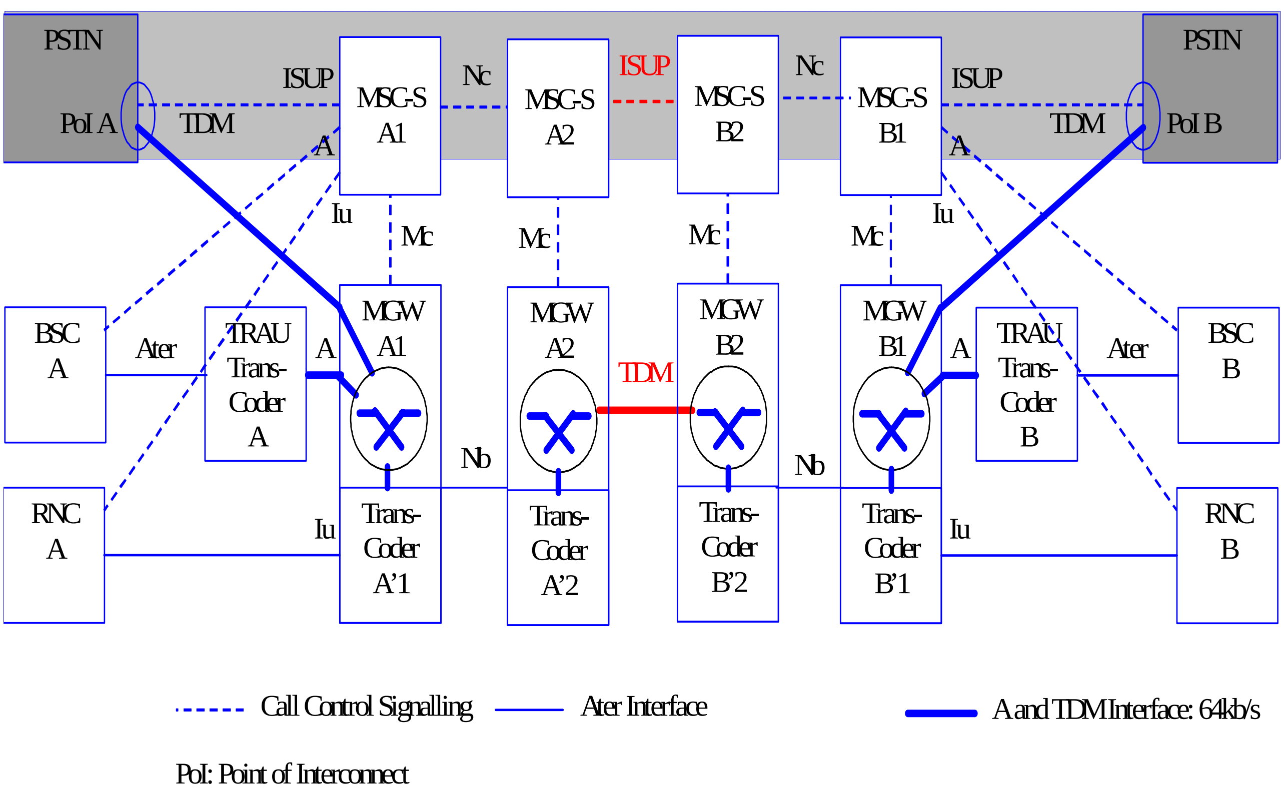


Figure 4.4-1 TDM Transit network between PLMNs from Release 4 onwards

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

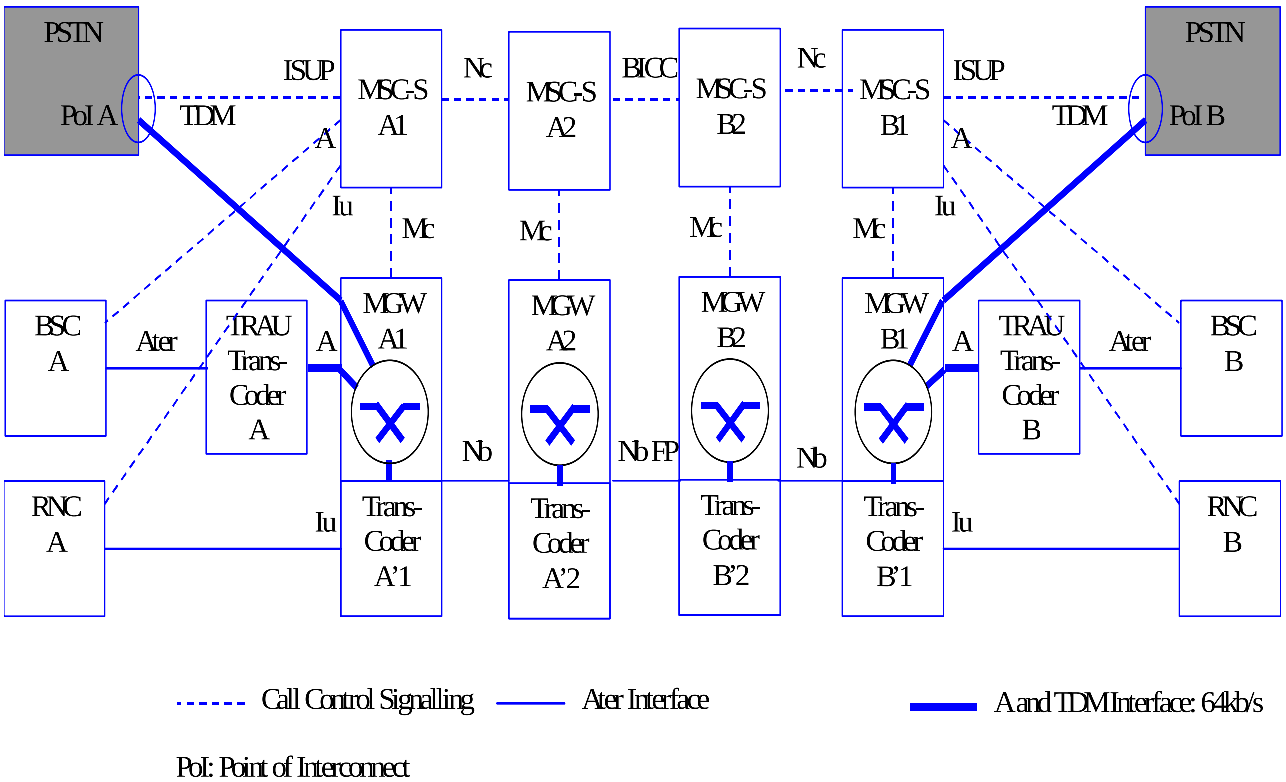


Figure 4.5-1 Packet Transport Transit network between PLMNs of REL5

This architecture shows two mobile Core Networks (BICNs) of Release 4 or 5 in layered architecture, with BICC and OoBTC on the Nc interface and speech in compressed form on the Nb interface. They are connected by a packet network, using the signalling and user plane protocols, which are used with the BICNs. All features as explained above for one BICN are of course valid inside each BICN and are not further reprinted here in all details. Indeed, in this scenario all features apply across network borders: OoBTC and TrFO can take place all along the path in the core networks, resulting in compressed speech with high bandwidth efficiency on all user planes for UE-to-UE calls even across PLMN borders.

[Editor's Note: Need to add a figure, which shows the BICN within the PLMNs and the connection using a packet transport with protocols different than the ones used within the PLMNs.]

5 Call Scenarios to be studied

[Editor's Note: In order to understand the typical call types and call mixes likely to be encountered in the networks covered by section 3 (above), it is proposed that a section be dedicated to documenting typical call scenarios. For Example the following scenarios should be studied:

A) Static cases (i.e. the scenario does not change after call setup)
A1) Call scenarios that require more than 1 transcoding stage
- MS-MS calls without TFO and TrFO
- MS-PSTN calls without TFO, but with bandwidth efficient transmission in the CN
-MS-MS calls with different codec types

A2) Call scenarios where compressed speech is carried inefficiently (i.e. in PCM frames)

B) Dynamic cases (i.e. where the scenario changes during the call)
B1) Call scenarios with changes at call setup
- Calls with cascades of TFO and TrFO
- Calls with call forwarding

B2) Call scenarios with changes due to handover:
- Call scenario with change of codec type due to intra GERAN handover
- Call scenario with change of codec type due to intra UTRAN handover
- Call scenarios where calls handover between 3G and 2G access technologies

C) Call Scenarios with CS domain to IMS Interworking]

The following call scenarios are those of interest for BARS functionality for the network scenarios detailed in clause 4. For each scenario the resources used in the MGWs and the bandwidth in use are described.

Resource utilisation in MGWs comprises the following aspects:

a) **Protocol Termination (PT)**. TFO requires the use of protocol handlers for the inband signalling.

b) **Re-framing (R)**. Depending on the scenario, it is necessary to reframe "the same bits of information", for example at transition from TDM to packet or vice versa. Reframing is also necessary where transcoding takes place.

c) **Transcoding (TC)**. Transcoding is needed to change from AMR to G.711 and vice-versa, but also needed to restore the PCM signal towards TDM networks at the end of a TrFO link.

d) **User Plane Termination (UP)**. Termination of the Iu/Nb User Plane Protocol.

The figures indicate the resources needed in the MGWs in various scenarios for the case where TFO is used (green) and the case where TrFO is used (blue). In addition to the resource utilisation also the bandwidth used is shown:

13 kbit/s:

64 kbit/s:

5.1 Mobile to Mobile Call Scenarios

5.1.1 BSC to RNC Call via BICN

UE A in the coverage area of a BSC connected via A interface to a MGW, calls UE B which is in the coverage area of a RNC connected via Iu interface to a different MGW. The call between the MGW's is carried via the Nb interface connecting them.

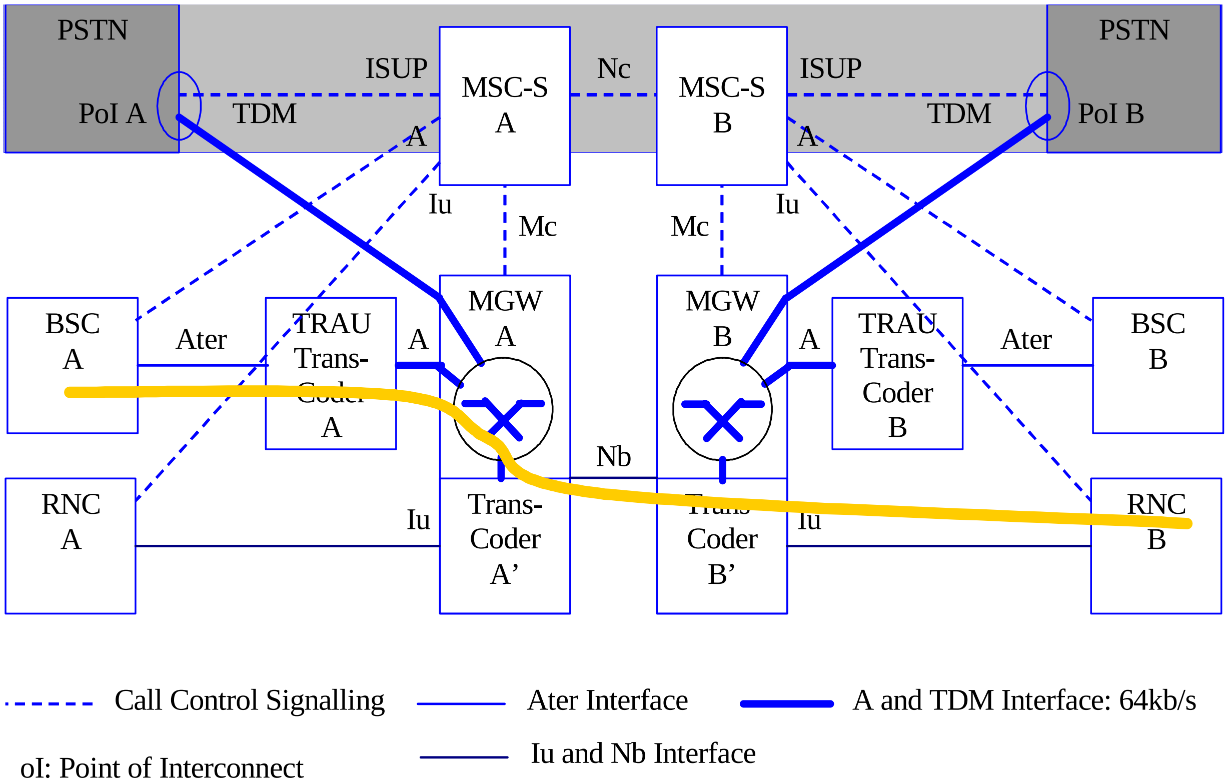


Figure 5.1.1-1 BSC to RNC Call via BICN

Harmonized OoBTC/TFO procedures allow for the interworking of TFO procedures on the A interface and OoBTC procedures on the Nb interface to provide for improved voice quality by removing unnecessary transcoding from the voice path, and high bandwidth efficiency on all portions of the speech path except the A interface between Transcoder A and MGW A in the figure.

1. MSC-S A and MSC-S B use OoBTC to minimize transmission bandwidth and the allocation of codecs on the speech path between MGW A and RNC B, using preferred BSC A Codec Configuration.

2. If TFO is supported but codec mismatch occurs then MSC-S A discovers the Codec Configuration at BSC A via MGW A using the TFO package on the Mc interface, it initiates TrFO/TFO codec negotiation harmonization if necessary, using OoBTC and/or TFO codec modification procedures.

3. MSC-S A establishes TFO operation between Transcoder A and MGW A.

4. When codec negotiation completes successfully, the Ater, Nb and Iu interfaces all carry compressed speech with high bandwidth efficiency. Transcoders A and A' remain in the speech path to perform necessary TFO protocol tasks and decoding but do not perform transcoding. Transcoder B' is not in the speech path.

The bandwidth and resource usage is analogous to sub-clause 5.1.3 below.

5.1.2 BSC to BSC Call via BICN

UE A in the coverage area of a BSC connected via A interface to a MGW, calls UE B which is in the coverage area of a BSC connected via A interface to a different MGW. The call between the MGW's is carried via the Nb interface connecting them.

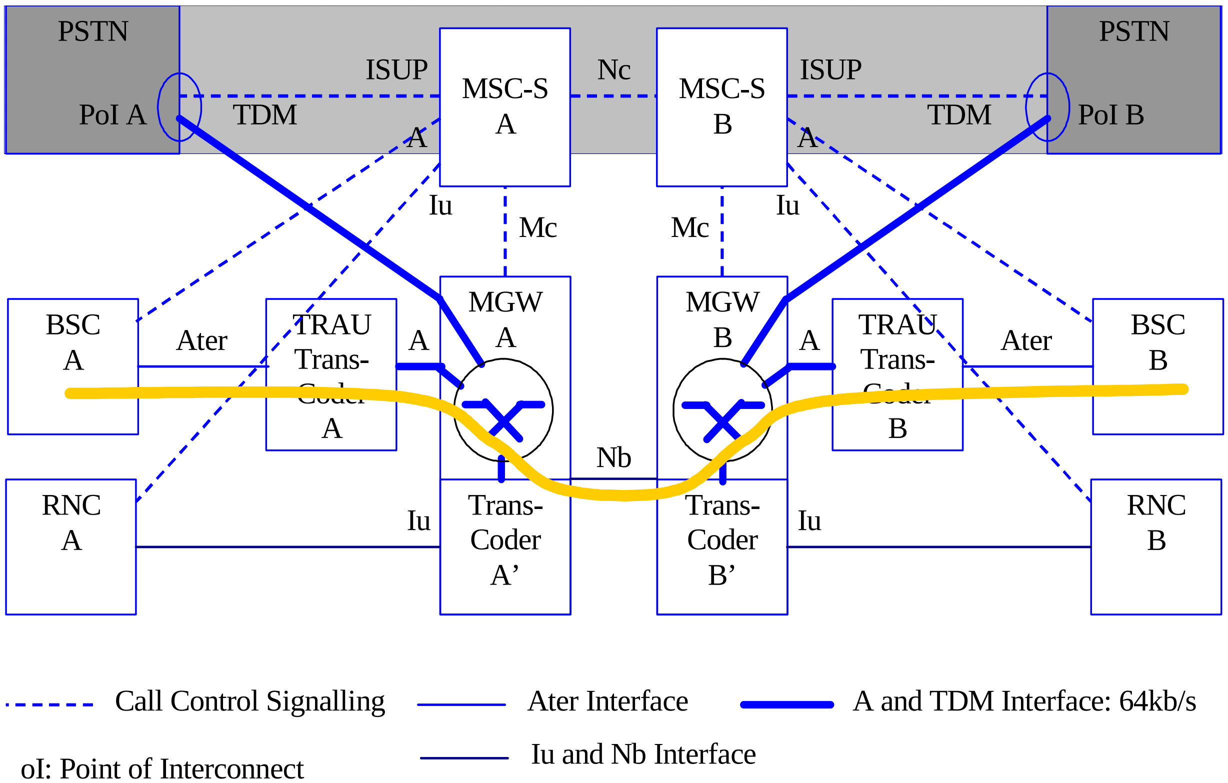


Figure 5.1.2-1 BSC to BSC Call via BICN

Harmonized OoBTC/TFO procedures allow for the interworking of TFO procedures on the A interfaces and OoBTC procedures on the Nb interface to provide for improved voice quality by removing unnecessary transcoding from the voice path, and high bandwidth efficiency on all portions of the speech path except the A interface between Transcoder A and MGW A and the A interface between MGW B and Transcoder B in the figure.

1. MSC-S A and MSC-S B use OoBTC to establish compressed speech over Nb between MGW A and MGW B.
2. If TFO is supported but codec mismatch occurs then the MSC-Ss discover the Codec Configurations at the BSCs via the MGWs using the TFO package on the Mc interface, they initiate TrFO/TFO codec negotiation harmonization, using OoBTC and/or TFO codec modification procedures.
3. When codec negotiation completes successfully, the Nb interface carries compressed speech with high bandwidth efficiency. Transcoders A, A', B' and B remain in the speech path to perform necessary TFO protocol tasks and decoding but do not perform transcoding.

Figure 5.1.2-2 shows the bandwidth and resources used in case TFO or TrFO are applied:

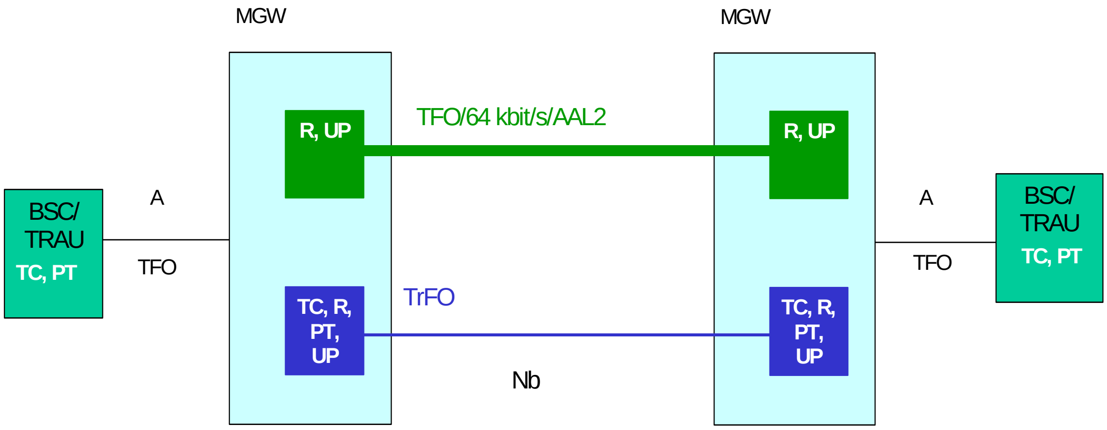


Figure 5.1.2-2 Bandwidth and resource usage for BSC to BSC Call via BICN

5.1.3 RNC to BSC Call via BICN

UE A in the coverage area of a RNC connected via Iu interface to a MGW, calls UE B which is in the coverage area of a BSC connected via A interface to a different MGW. The call between the MGW's is carried via the Nb interface connecting them.

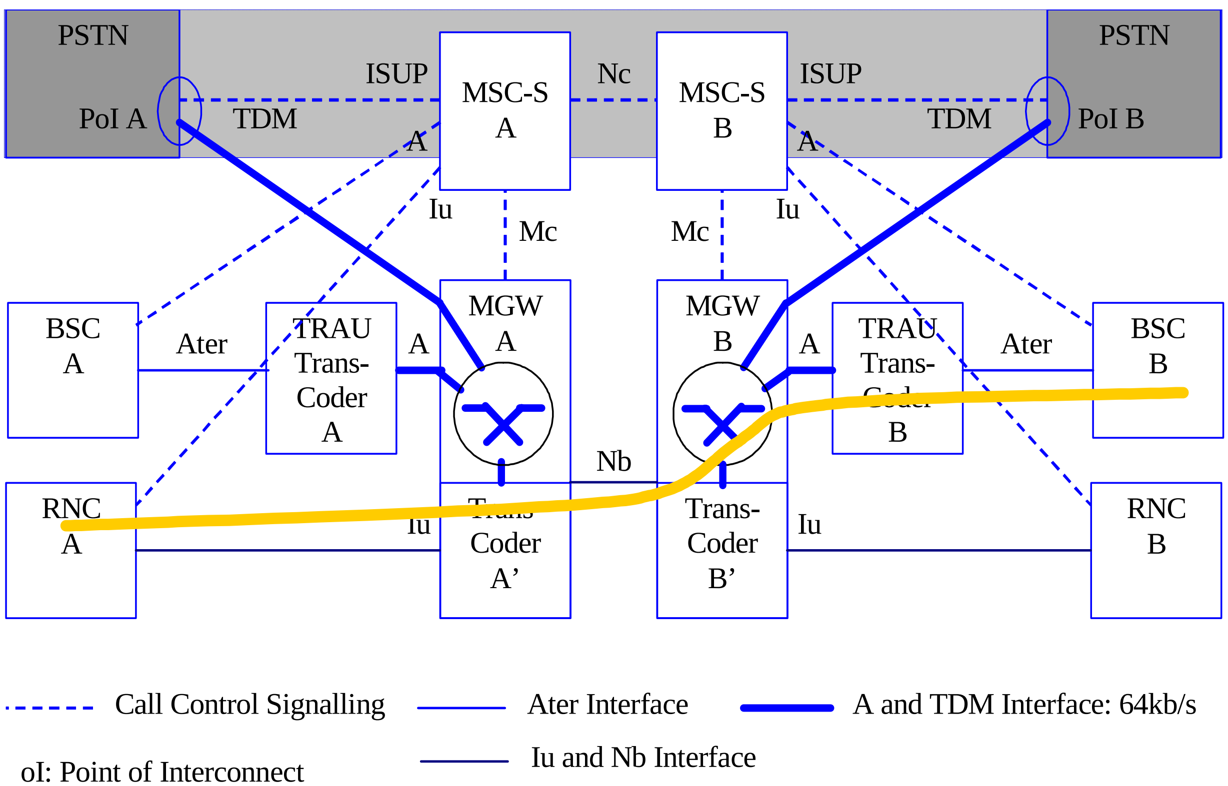


Figure 5.1.3-1 RNC to BSC Call via BICN

Harmonized OoBTC/TFO procedures allow for the interworking of OoBTC procedures on the Nb interface and TFO procedures on the A interfaces to provide for improved voice quality by removing unnecessary transcoding from the voice path, and high bandwidth efficiency on all portions of the speech path except the A interface between MGW B and Transcoder B in the figure.

1. MSC-S A and MSC-S B use OoBTC to minimize transmission bandwidth and the allocation of codecs on the speech path between RNC A and MGW B, using assumed information about the BSC B Codec Configuration.

2. If TFO is supported but codec mismatch occurs then MSC-S B discovers the Codec Configuration at BSC B via MGW B using the TFO package on the Mc interface, it initiates TrFO/TFO codec negotiation harmonization if necessary, using OoBTC and/or TFO codec modification procedures.

3. MSC-S B establishes TFO operation between MGW B and Transcoder B.

4. When codec negotiation completes successfully, the Ater, Nb and Iu interfaces all carry compressed speech with high bandwidth efficiency. Transcoder A' is not in the speech path. Transcoders B' and B remain in the speech path to perform necessary TFO protocol tasks and decoding but do not perform transcoding.

Figure 5.1.3-2 shows the bandwidth and resources used in case TFO or TrFO are applied:

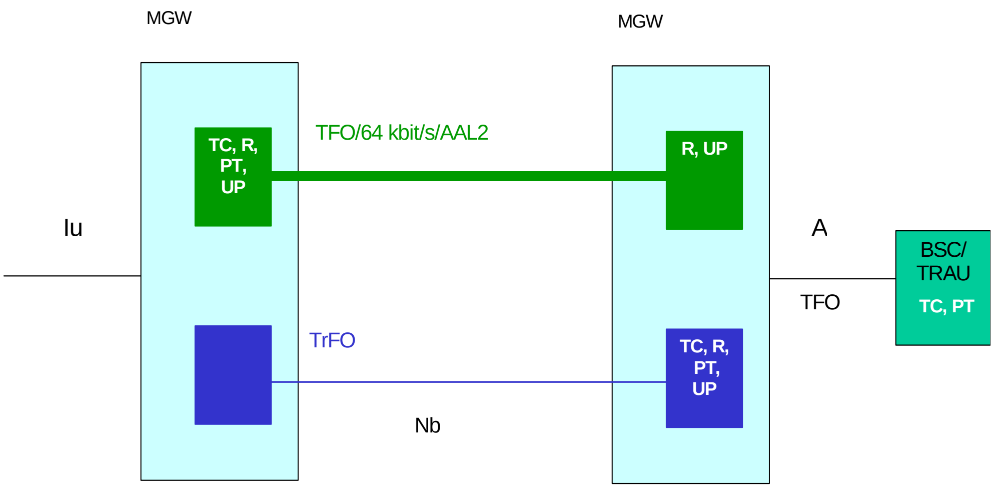


Figure 5.1.3-2 Bandwidth and resource usage for RNC to BSC Call via BICN

5.1.4 RNC to RNC Call via BICN

UE A in the coverage area of a RNC connected via Iu interface to a MGW, calls UE B which is in the coverage area of a RNC connected via Iu interface to a different MGW. The call between the MGW's is carried via the Nb interface connecting them.

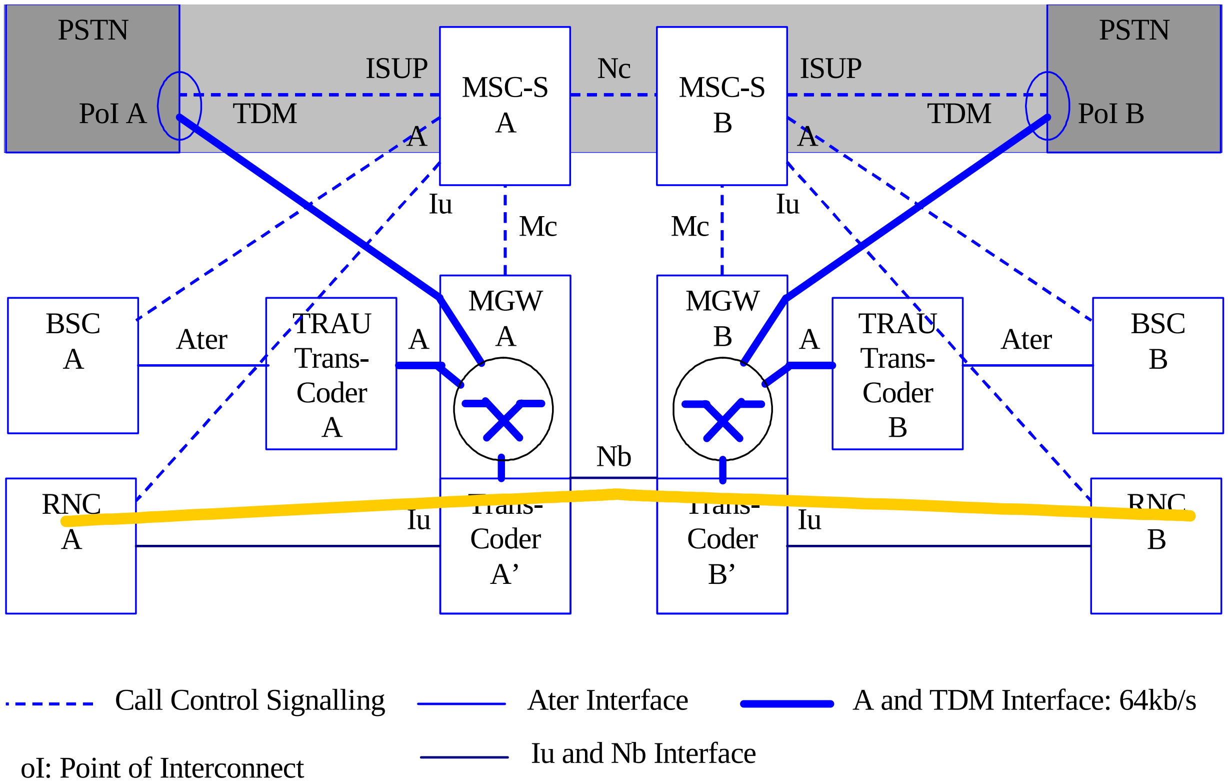


Figure 5.1.4-1 RNC to RNC Call via BICN

OoBTC procedures associated with the Nb interface provide for improved voice quality by removing unnecessary transcoding from the voice path, and high bandwidth efficiency on all portions of the speech path in the figure.

1. MSC-S A and MSC-S B use OoBTC to minimize transmission bandwidth and the allocation of codecs on the speech path between RNC A and RNC B.

2. When codec negotiation completes successfully, the Nb and Iu interfaces all carry compressed speech with high bandwidth efficiency. Neither Transcoder A' nor Transcoder B' remains in the speech path.

Figure 5.1.4-2 shows the bandwidth and resources used in case TFO or TrFO are applied:

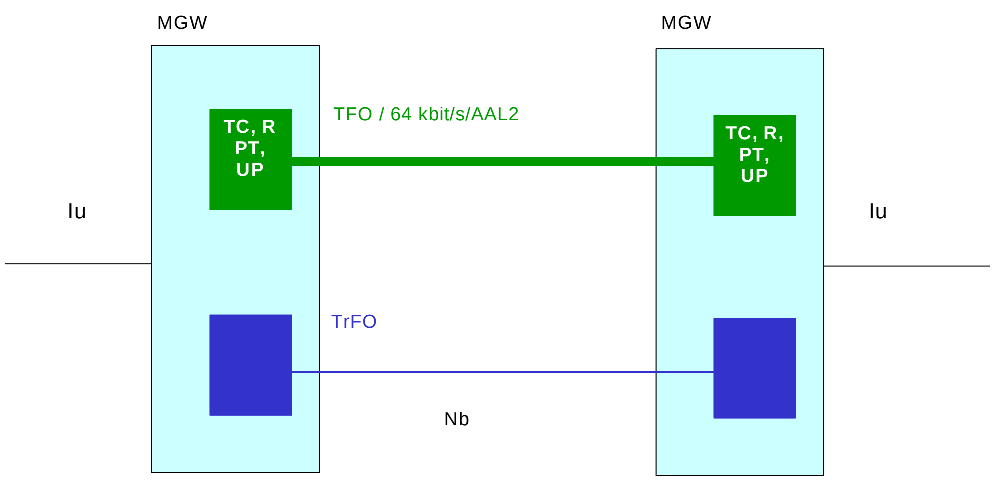


Figure 5.1.4-2 Bandwidth and resource usage for RNC to RNC Call via BICN

5.2 Mobile to PSTN Call Scenarios

5.2.1 BSC to PSTN Call via BICN

UE A in the coverage area of a BSC connected via A interface to a MGW, calls PSTN phone B which is in a switch connected via a TDM interface to a different MGW. The call between the MGW's is carried via the Nb interface connecting them.

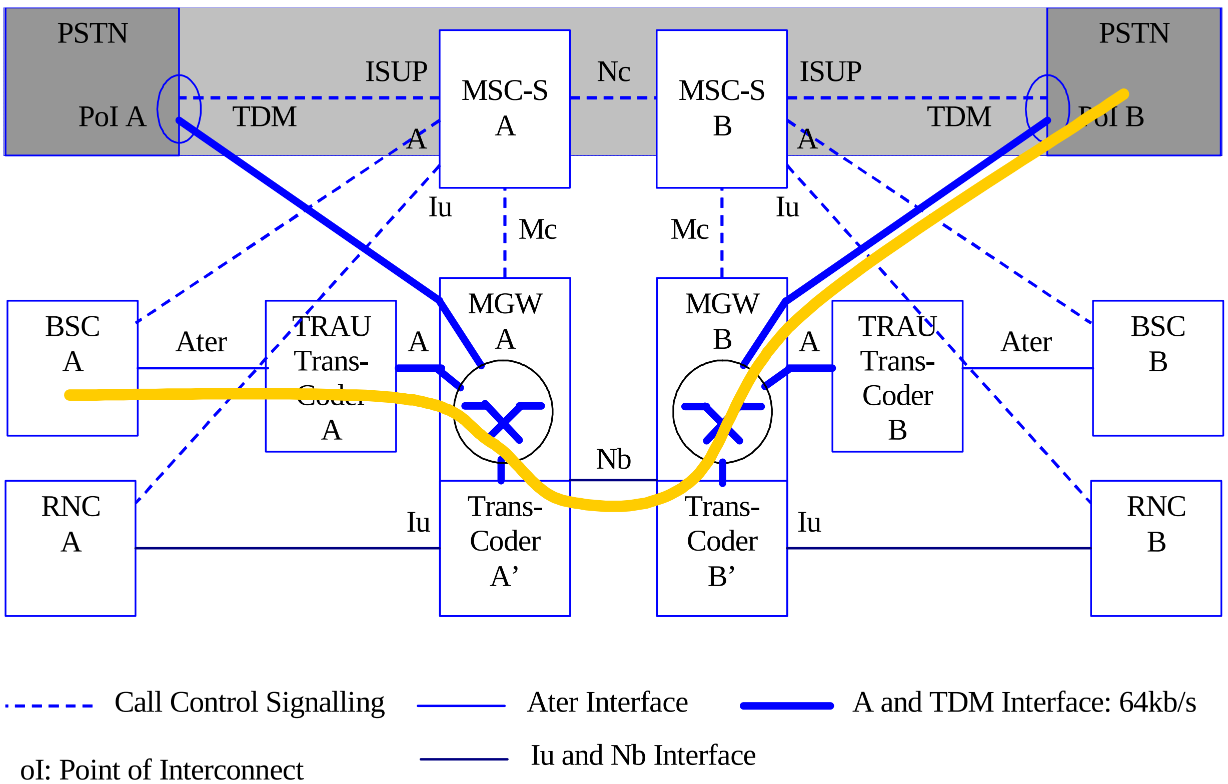


Figure 5.2.1-1 BSC to PSTN Call via BICN

Harmonized OoBTC/TFO procedures allow for the interworking of TFO procedures on the A interface and OoBTC procedures on the Nb interface to provide for high bandwidth efficiency on all portions of the speech path except the A interface between Transcoder A and MGW A and the TDM path to the PSTN in the figure.

1. MSC-S A and MSC-S B use OoBTC to establish compressed speech over Nb between MGW A and MGW B.
2. If TFO is supported on the A interface but codec mismatch occurs then MSC-S A discovers the Codec Configuration at BSC A via MGW A using the TFO package on the Mc interface, it initiates TrFO/TFO codec negotiation harmonization, using OoBTC and/or TFO codec modification procedures.
3. When codec negotiation completes successfully, the Nb interface carries compressed speech with high bandwidth efficiency. Transcoders A and A' remain in the speech path to perform necessary TFO protocol tasks and decoding but do not perform transcoding. Transcoder B' performs the necessary transcoding to PCM.

Figure 5.2.1-2 shows the bandwidth and resources used in case TFO or TrFO are applied:

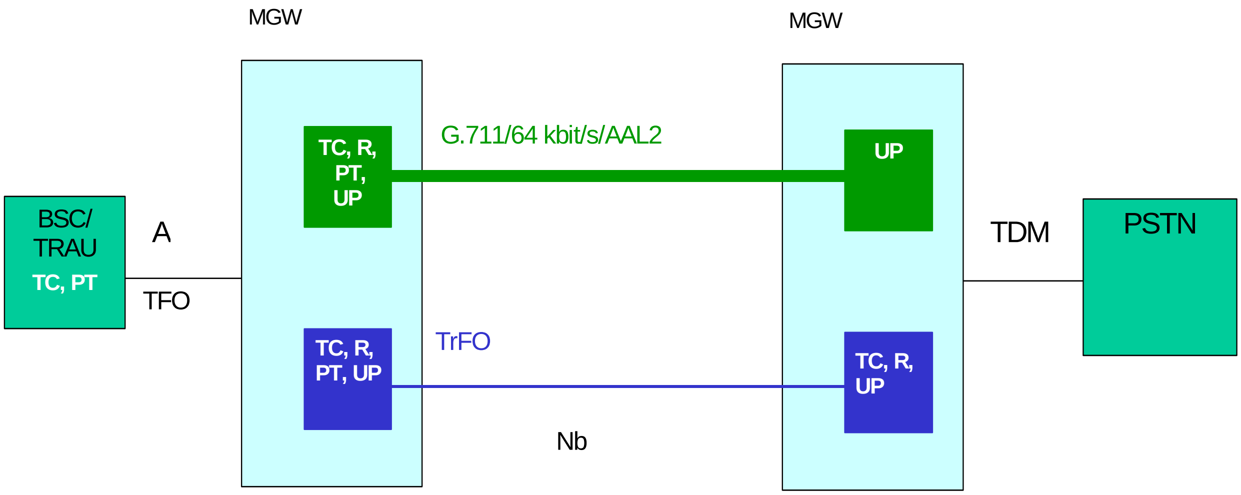


Figure 5.2.1-2 Bandwidth and resource usage for BSC to PSTN Call via BICN

5.2.2 RNC to PSTN Call via BICN

UE A in the coverage area of a RNC connected via Iu interface to a MGW, calls PSTN phone B which is in a switch connected via a TDM interface to a different MGW. The call between the MGW's is carried via the Nb interface connecting them.

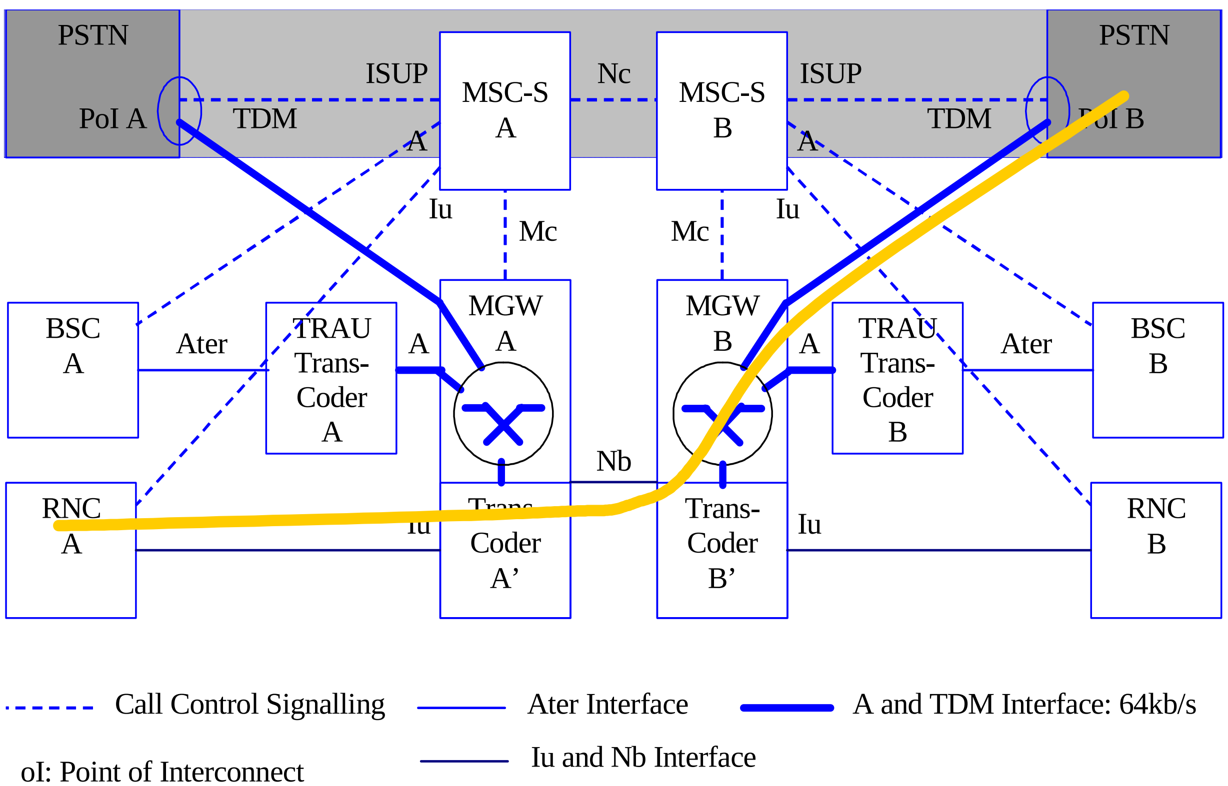


Figure 5.2.2-1 RNC to PSTN Call via BICN

1. OoBTC procedures associated with the Nb interface provide for high bandwidth efficiency on all portions of the speech path except the TDM path to the PSTN in the figure. MSC-S A and MSC-S B use OoBTC to

minimize transmission bandwidth and the allocation of codecs on the speech path between RNC A and PSTN B.

2. When codec negotiation completes successfully, the Nb and Iu interfaces carry compressed speech with high bandwidth efficiency. Transcoder A' is not in the speech path. Transcoder B' performs the necessary transcoding to PCM.

Figure 5.2.2-2 shows the bandwidth and resources used in case TFO or TrFO are applied:

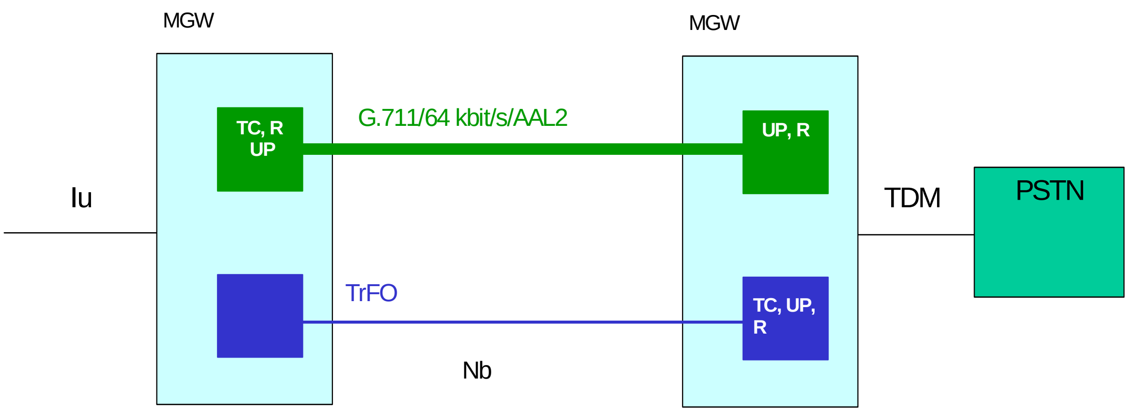


Figure 5.2-2 Bandwidth and resource usage for RNCC to PSTN Call via BICN

5.3 Roaming and Multi-Network Call Scenarios

5.3.1 BSC (HPLMN) to BSC (VPLMN) Call

UE A in the coverage area of a BSC connected via A interface to a MGW in the HPLMN, calls UE B which is roaming in the coverage area of a BSC connected via A interface to a MGW on the VPLMN. The call from the HPLMN MGW's is carried via TDM circuits to a Gateway MGW in the UE B's HPLMN and further on to a Gateway MGW in the VPLMN, which then routes the call to the destination MGW (the one connected to the BSC) over the Nb interface.

The bandwidth and resource usage are analogous to sub-clause 5.2.1 for the HPLMN A and VPLMN. The MGW in HPLMN B does not need to perform Protocol Termination, Transcoding, Re-framing, or User Plane Termination.

Note: in the above it is assumed that the TDM links between networks use DCMEs and thus TFO is not applicable across them.

Editor's note: need to include the scenarios where TFO can be used on the TDM link.

5.3.2 BSC (HPLMN) to RNC (VPLMN) Call

UE A in the coverage area of a BSC connected via A interface to a MGW in the HPLMN, calls UE B which is roaming in the coverage area of a RNC connected via Iu interface to a MGW on the VPLMN. The call from the HPLMN MGW's is carried via TDM circuits to a Gateway MGW in the UE B's HPLMN and further on to a Gateway MGW in the VPLMN, which then routes the call to the destination MGW (the one connected to the RNC) over the Nb interface.

The bandwidth and resource usage are analogous to sub-clause 5.2.1 for the HPLMN A and to sub-clause 5.2.2 for the VPLMN. The MGW in HPLMN B does not need to perform Protocol Termination, Transcoding, Re-framing, or User Plane Termination.

Note: in the above it is assumed that the TDM links between networks use DCMEs and thus TFO is not applicable across them.

Editor's note: need to include the scenarios where TFO can be used on the TDM link.

5.3.3 RNC (HPLMN) to BSC (VPLMN) Call

UE A in the coverage area of a RNC connected via Iu interface to a MGW in the HPLMN, calls UE B which is roaming in the coverage area of a BSC connected via A interface to a MGW on the VPLMN. The call from the HPLMN MGW's is carried via TDM circuits to a Gateway MGW in the UE B's HPLMN and further on to a Gateway MGW in the VPLMN, which then routes the call to the destination MGW (the one connected to the BSC) over the Nb interface.

The bandwidth and resource usage are analogous to sub-clause 5.2.2 for the HPLMN A and to sub-clause 5.2.1 for the VPLMN. The MGW in HPLMN B does not need to perform Protocol Termination, Transcoding, Re-framing, or User Plane Termination.

Note: in the above it is assumed that the TDM links between networks use DCMEs and thus TFO is not applicable across them.

Editor's note: need to include the scenarios where TFO can be used on the TDM link.

5.3.4 RNC (HPLMN) to RNC (VPLMN) Call

UE A in the coverage area of a RNC connected via Iu interface to a MGW in the HPLMN, calls UE B which is roaming in the coverage area of a RNC connected via Iu interface to a MGW on the VPLMN The call from the HPLMN MGW's is carried via TDM circuits to a Gateway MGW in the UE B's HPLMN and further on to a Gateway MGW in the VPLMN, which then routes the call to the destination MGW (the one connected to the RNC) over the Nb interface.

The bandwidth and resource usage are analogous to sub-clause 5.2.2 for the HPLMN A and for the VPLMN. The MGW in HPLMN B does not need to perform Protocol Termination, Transcoding, Re-framing, or User Plane Termination.

Note: in the above it is assumed that the TDM links between networks use DCMEs and thus TFO is not applicable across them.

Editor's note: need to include the scenarios where TFO can be used on the TDM link.

6 General Requirements for Architectural Solutions

[Editors Note: It is proposed that the TR should cover general requirements such as backwards compatibility and interworking with and evolution of existing standard solutions (TrFO/TFO) etc.]

- Work between PLMNs (where agreements and intervening networks permit).

- Interworking fully defined with existing 3GPP standards (e.g. TrFO, TFO)
- Support for Interworking with IMS
- Backward compatible with existing GSM (R99) Radio Access networks.
- Backward compatible with existing terminals
- Does not require implementation of non-standard interfaces on the Media Gateway (e.g. Ater).
- Support for Local Lawful Intercept requirements

7 Requirements and Architectural Solutions for Resource Savings

[Editor's Note: An objective of the work item is to identify the full set of requirements and assess the architectural solutions for resource savings, with specific consideration to networks supporting A/Gb mode and the BICN. It is proposed that the TR studies the following topics.

Comparison of resources needed for in band and out of band transcoder control and codec negotiation/initialisation/modification. Those resources are in terms of equipment and transactions needed to set up calls and deal with any modifications of these calls.

Resource requirements for interworking Nb interface with wireline networks (PSTN or Transit networks) with TDM, ATM or IP transport.]

- Reduce the total number of transcoding equipment in a A/Gb mode network using R4 core network architecture. This is very important for growing A/Gb mode networks.

8 Requirements and Architectural Solutions for Bandwidth Savings

[Editor's Note: An objective of the work item is to identify the full set of requirements and assess the architectural solutions for bandwidth savings, with specific consideration to networks supporting A/Gb mode and the BICN. It is proposed that the TR investigates the following topics.

The bandwidth usage efficiency of the Nb interface for carrying either compressed speech or G.711 with ATM transport or IP transport

The bandwidth usage efficiency of the A interface for carrying compressed speech.]

8.1 Background

The 3GPP architecture must support bandwidth usage efficiency on the most highly utilized user plane interfaces in the system. Candidates for optimisation include:

- The path between RNC and MGW. This is the Iu interface and is already optimised.
- The path between MGWs within a BICN. When this path uses a packet network, it is the Nb interface and is optimised for TrFO.

- The path between BSC and MGW. This is a combination of the Ater and A interfaces via the TRAU. The Ater interface is already optimised but the A interface uses 64 kbps facilities on a TDM interface. In this form, the A interface is not a candidate for bandwidth optimisation, but is consistent with TFO.
- The path between MGWs in different PLMNs. This interface is typically TDM and may be consistent with TFO if no DCMEs or other non-TFO IPE (e.g. echo cancellers) are in the path. If a packet transport network is available between the PLMNs, then OoBTC may be applicable.

Other sections describe how harmonized OoBTC and TFO procedures enable some combination of TrFO, TFO and transcoding at the edge in various scenarios involving media flow on these paths. The A and TDM interfaces do not yet support the same degree of bandwidth usage efficiency as the Ater, Iu and Nb interfaces.

If a path between MGWs within a BICN does not already use TrFO over an Nb interface, it can be optimised by doing so. The path between MGWs may used TFO. The path between PLMNs may also support either TFO or TrFO. If a path between BSC and MGW is significantly comprised of an A interface, no standard method exists for realizing higher bandwidth usage efficiency on this portion of the path. The next section includes discussion of two alternative architectures to address this issue.

8.2 Requirements

- Reduce bandwidth requirements for A/Gb mode traffic in the packet transport network between Media Gateways across Nb interface.
- Enact transcoding at the edge of the network for calls to PSTN or other incompatible networks.

8.3 Architectural Solutions

8.3.1 A-ter interface to the MGW

Editor's Note: The diagrams and text needs to be extended to cover the signalling aspects

Editor's Note: The TR needs to be restructured so that all the Ater related text is gathered together in one place.

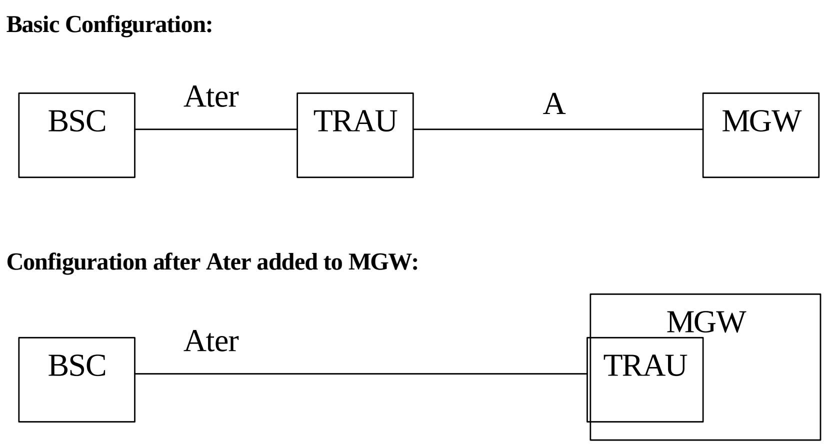


Figure 8.3.1-1 BSC to MGW path before and after adding Ater interface to MGW

Figure 8.3.1-1 depicts the BSC to MGW path as it would appear if an Ater interface is standardized for the MGW, and the TRAU function is performed within the MGW. This corresponds to the functional distribution RNC and MGW across the Iu interface. The advantage of this configuration is that all scenarios described herein using Nb packet transport between MGWs can support end-to-end OoBTC procedures for TrFO or transcoder at the edge since there is no need to perform TFO on any interface. Harmonized OoBTC/TFO procedures support all other scenarios described herein that include this configuration option for the BSC to MGW path, may include Nb packet transport between some MGWs in the path, and include at least one TDM interface between some pair of MGWs in the path, e.g., between PLMNs. But in the case there is TC at the edge of the network with TDM transit network to another TrFO network then TFO on this transit network will improve speech quality.

8.3.2 MGW collocated with TRAU

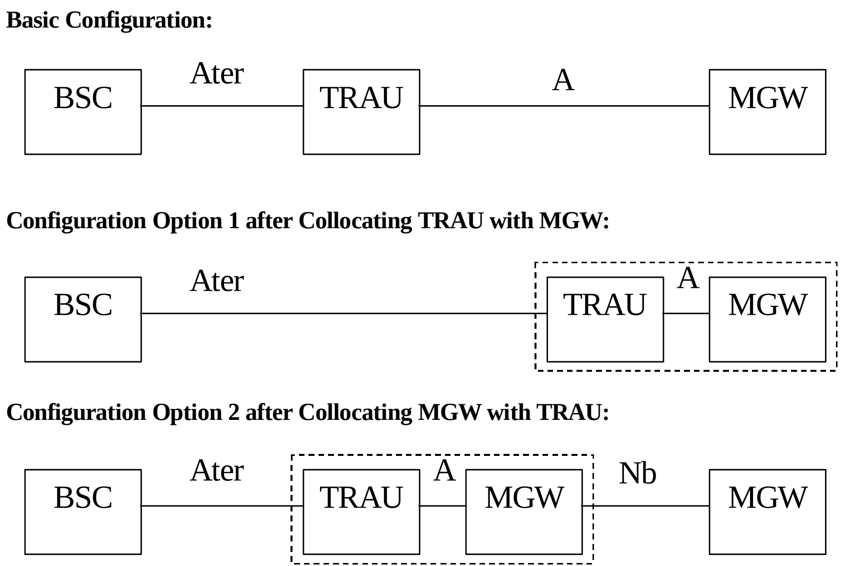


Figure 8.3.2-1 Options for BSC to MGW path after collocating MGW with TRAU

Figure 8.3.2-1 depicts two options for how the BSC to MGW path would appear if either the TRAU is moved to be collocated with (in physical proximity to) the MGW, or a new MGW is introduced and collocated with the TRAU. Either approach shortens the portion of the BSC to MGW path comprising an A interface to a negligible portion of the overall path. Harmonized OoBTC/TFO procedures support all scenarios described herein that includes either of these configuration options for the BSC to MGW path.

9 Requirements and Architectural Solutions for Speech Quality Improvements

9.1 Requirements

[Editor's Note: An objective of the work item is to identify the full set of requirements and assess the architectural solutions for improved speech quality, with specific consideration to networks supporting A/Gb mode and the BICN. It is proposed that the TR investigates the following topics.

Requirements for support of Wideband AMR

Requirements for reduction of transcoding stages

Requirements for speech quality enhancement features]

The general requirement to ensure maximum speech quality that TrFO and TFO attempt to meet is to prevent unnecessary transcoding on mobile-to-mobile and mobile to fixed network. TSG SA WG4 has studied the degradation to speech quality for mobile to mobile calls with tandeming of speech codecs, and the following is a summary of conclusions from the reports 3GPP TR 26.975 [6] and 3GPP TR 26.976 [7].

Section 7 of 3GPP TR 26.975 [6] shows that tandeming tests were conducted by SA4 in the past and they showed that the degradation in single tandeming compared to TFO between EFR and AMR12.2 is not significant to the user. To quote: "Tandeming with the clean speech error free 12.2 and 10.2 modes of AMR do not significantly degrade the single encoding performances of any of the AMR codec (modes) or existing GSM codecs." Transcoding between FR and AMR, however, does introduce degradation.

9.2 Architectural Solutions

9.2.1 Mobile to Mobile Calls Scenarios: 5.1.1 and 5.1.3

For mobile to mobile calls between a BSC and an RNC within a PLMN, the call scenarios in section 5 show that the existing architecture can employ TFO on the A interface supporting AMR-WB / AMR / EFR / FR / HR, andcompressed speech (TrFO) on the Iu interface supporting AMR-WB / AMR. If compatible codecs are available, then the network need not perform any transcoding. OoBTC can be used within the BICN and TrFO be established between the MGWs, thus achieving optimal bandwidth saving with optimal voice quality. If compatible codecs are not available, a single transcoding point at the MGW then exists between GSM on one side and UTRAN on the other and, therefore, the requirement to prevent unnecessary transcoding in order to not perceptibly degrade the speech quality is fulfilled. Again the BICN can operate in TrFO to achieve optimal bandwidth savings.

For BSC to BSC calls via a BICN, the network should not need to perform any transcoding when the mobiles share at least one common codec. In some cases, e.g. if only one side supports AMR in multi-mode configuration and the other side supports only single-mode codecs (FR / HR / EFR), it needs to be considered whether to perform a single transcoding in order to gain optimal voice quality. In such cases, the radio error robustness of the AMR may be more important than the TFO connection of a single-mode codec. For details see TS 28.062, section F.3 [4].

Whether a transcoding-free link can be established through the BICN (for BSC - BSC or BSC – RNC calls) depends on the codec types used for TFO. If at least one side supports AMR, or AMR-WB or EFR, then optimal bandwidth saving and optimal voice quality can be achieved. Otherwise a 64kbps transparent PCM channel with TFO has to be established through the BICN for optimal voice quality. The reason is that up to 3GPP REL5 no Nb framing is specified for FR and HR. Also, though Nb (and Iu) framing for EFR is supported according to TS 26.103 [9], the specification in TS 26.102 [8] does not seem complete.

9.2.2 Mobile to PSTN Calls: Scenarios 5.2.x

For GSM/UMTS mobile to PSTN calls, the existing solutions allow for a single, necessary, transcoding point within the MGW at the edge of the BICN, close to the point of interconnect.

9.3 Summary

Architectural solutions exist for the common interworking scenarios of mobile to mobile/PTSN calls, which avoid or minimise degradation of speech quality. Optimal Voice Quality together with Optimal Bandwidth Saving is possible for AMR, AMR-WB and EFR. For EFR the Nb (and Iu) framing does not seem completely specified and for FR and HR the optimal bandwidth saving would need the Nb framing for these codec types to be specified. A useful output of this TR would be for these (or at least EFR) be standardised for Nb framing in order for a more optimal voice quality to be accomplished.

10 Requirements and Architectural Solutions for Avoiding Duplication in Transcoder Development

10.1 Background and Requirements

The GSM and UMTS systems will co-exist for many years. Most operators have significantly less UMTS spectrum than they have GSM spectrum, so, operators have to optimise their utilisation of the combined spectrum pool.

Hence the introduction of a new speech coder (particularly one that is best suited for mobile to mobile calls) requires support for that codec in both 2G and 3G coverage areas.

Note: over GSM's 12 years, 4 new speech coders have been developed (HR, EFR, AMR, WB-AMR). Hence we can anticipate that further speech coders will appear at the rate of about 1 every 3 years.

With the current architecture this requires both TRAUs in the 2G BSS and Transcoders in 3G MSCs/Media GateWays to be developed and installed.

This has many disadvantages, eg:

a) duplicated development cost

b) the feature is difficult to use until the slowest of MGW and TRAU development is finished

c) if/when GSM is decommissioned, TRAUs in the BSS will probably have to be discarded

Hence it is required to consider how a graceful migration of transcoding functionality from BSS to MGW can be achieved.

10.2 Architectural Solutions

10.2.1 A-ter interface to the MGW

If new Transcoders are only implemented on the MGW, then the MGW will need to be able to be connected to GSM BTSes (via the BSC). Given that there is a very large installed base of GSM base stations but only a limited installed base of MGWs, it seems logical that the MGW adapts itself to handle the existing interface to the BTS.

On the user plane, this interface is defined in TS 28.060/28.061. Given that multi-vendor interoperability is required for TFO, and, the TFO standard (TS 28.062) is closely related to TS 28.060/28.061, it seems reasonable to assume that TS 28.060/28.061 are (or can be made into) open standards.

Many BSC vendors support TRAUs located at the MSC site. The O+M for these remote TRAUs is generally regarded as proprietary. However, if the transcoding is located within the MGW, then the MGW O+M can be used for this task.

For the A interface control plane, the MSC already controls the allocation of the circuit. Further study of this is required, in particular for the Mc interface.

10.2.2 Other architectures

For Further Study.

11 Conclusions

Architectural solutions exist for the common interworking scenarios of mobile to mobile/PSTN calls, which avoid or minimise the degradation of speech quality.

Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2003-08					First draft of TR – creation of version 0.0.0 at TSG SA2 #33	---	0.0.0
2003-08					TR updated to include contribution from SA2#33	0.0.0	0.1.0
2003-08					Typographical error corrected	0.1.0	0.1.1
2003-08					TR Updated to include contributions from SA2#34	0.1.1	0.2.0
2003-11					TR Updated to include contributions from SA2#35	0.2.0	0.3.0
2003-12					TR Updated to include contributions from SA2#36	0.3.0	0.4.0
2003-12					First presentation for Information	0.4.0	1.0.0