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Summary and Recommendation

Tandem Free Operation (TFO) and Transcoder Free Operation (TrFO) should smoothly interact and the differences for the (radio) access networks and the terminals should be hidden completely or to a very large extend. But this is only possible, if both procedures are harmonised.
It is therefore recommended to establish a much closer formal relationship between all involved groups within 3GPP and also to other bodies (TIA, 3GPP2, …). It seems realistic that both TFO and TrFO will be finalised in release 2000 for GSM and UMTS.

1. Introduction

<u>Tandem Free Operation (TFO)</u> is an inband signalling procedur in many mobile systems (defined in GSM, TDMA, CdmaOne and Cdma2000, foreseen for UMTS release 2000). For optimal speech quality TFO avoids tandem (double) speech compression in case of mobile-to-mobile calls. In its basic form TFO involves only the transcoding equipment (TRAU / TC) in the fixed part of the network. TFO establishes a virtually transparent digital channel between the two terminals and transmits the compressed speech untouched within the core network (CN). But it still needs these TRAU equipments for call setup and fast fall_back to normal mode. TFO can not resolve Codec_Type mismatch situations without the support of out-of-band signalling. TFO is the method of choice when the call control layer within the core network is not aware of speech compression, respectively assumes G.711 as default.

<u>Transcoder Free Operation (TrFO)</u> is an out-of-band signalling procedure foreseen in ITU and UMTS to achieve the same speech quality enhancement as TFO. But is allocates the transcoding equipment only when needed (thus it saves hardware resources) and it optimises the transport capacity in the fixed part of the network (in case of AMR it can be about one order of magnitude more bandwidth efficient). The call control layer of the CN needs to understand fully the implications of TrFO, especially for all kind of supplementary services (like DTMF and other signalling tone insertion, call forwarding, announcements, conference bridges, etc.) and all kinds of In-Path_Equipments, such as echo cancellers and level shifters.

BUT: Once the call is established in TFO or TrFO, the traffic handling in TFO and TrFO is very similar and interconnection between TFO and TrFO is possible, if the solutions are harmonised. Especially important aspects are:

- negotiation at setup of TrFO and TFO for Codec_Type and Codec_Attributes
- modification of Codec_Type and Codec_Attributes during the call
- handling of handover and other TrFO/TFO interruption (TrFO break)
- compatible frame formats (contents) for the traffic transport
- compatible inband signalling for rate control, discontinuous transmission and bad frame handling.

Up to now the standardisation work for TFO and TrFO was not strongly coordinated due to high time pressure and due to many open questions on fundamental issues. But now the time has come to finalise both procedures in a harmonised way.
The design of T(r)FO should take the interaction with other systems into account.

2. General Principle of Tandem Free Operation

Figure 1 (extract from GSM 08.62) shows the principle of a TFO architecture. The yellow parts are the only TFO specific extensions (TFO_Protocol and its associated UL_ and DL_ TFO processes).

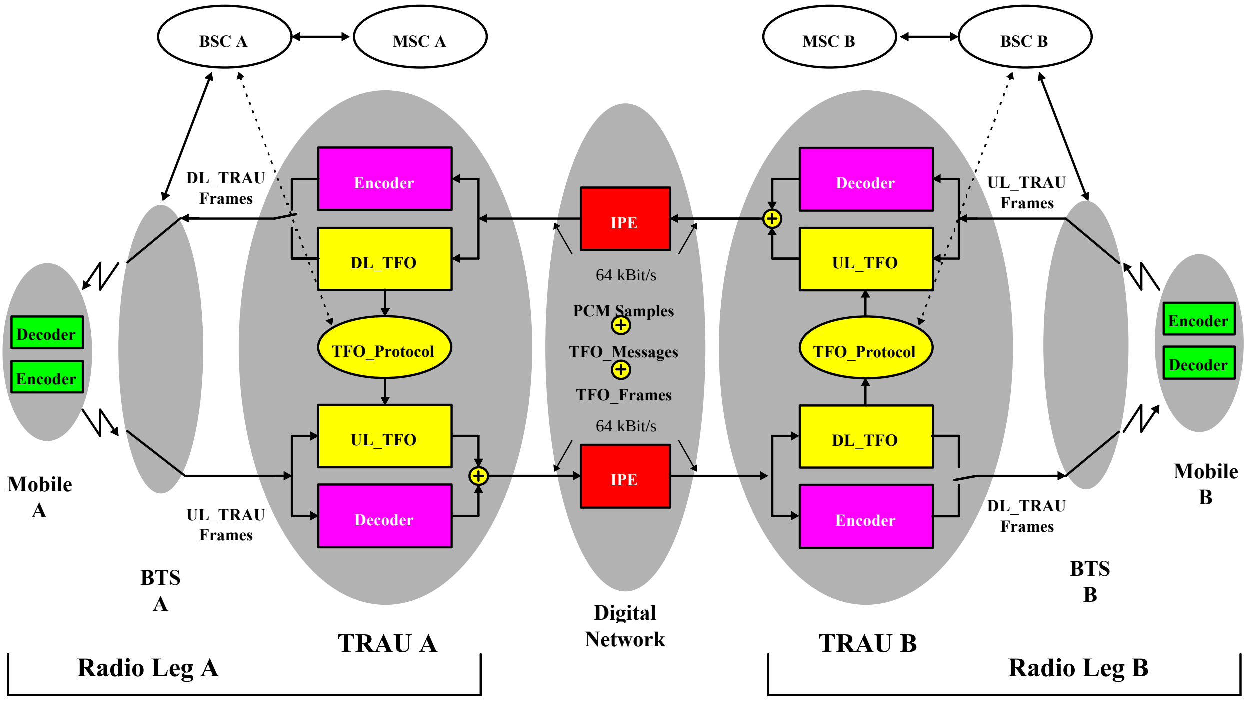

Figure 1: Functional Entities for Handling of Tandem Free Operation in MS-MS calls

2.1 Call Setup and TFO Setup

The call is set up by the Core Network (MSCs) without any knowledge about TFO. Also the BSCs have only limit knowledge of TFO. In case of a mobile-to-mobile connection the call is therefore at the beginning in “normal” or “tandem” mode: the speech signal is encoded in Mobile A, decoded in TRAU A, transported through a unknown digital (or analogue) transport network, encoded again in TRAU B and decoded finally in Mobile B. Thus the speech is coded twice and transmitted in 64 kBit/s channels between the TRAUs.

But each TRAU knows about its own capabilities: System_Identification, Codec Type and Codec_Attributes and, if supported by the BSC, it knows the Codec_List of alternative Codec_Types and Codec_Attributes.

If enabled, the TRAUs try to set up a TFO connection through the established traffic channel by sending a hidden inband signal and waiting for proper response. This TFO inband signalling steals bits from the speech traffic channel and is therefore not very fast (500 Bit/s). If a suitable TFO_partner (other TRAU or other compatible device) is found and the path between the TFO_Partners is transparent, they insert and extract the speech parameters into/from the LSBs of the PCM signal. Thus the speech quality improves, the transport channel remains unchanged at 64 kBit/s. Both TFO_Partners monitor permanently the incoming traffic stream and therefore can fall back to normal tandem connection very quickly without intervention of other (core) network entities. All supplementary services can remain unchanged, but when invoked the TFO is broken for that duration.

Typically TFO can only established between exact identical Codec_Types with identical Codec_Attributes. For most existing Codec_Types (e.g. GSM_FR, GSM_EFR, TDMA_FR, TDMA_EFR, PDC_FR, PDC_EFR … to mention just some) this is rather simple and TFO for these is considered solved in standardisation.

Both TFO partners firstly exchange the active Codec_Type and its Codec_Attributes. If these match, then TFO is established immediately, in typical call scenarios about 1s after the speech path is through-connected.

TFO takes care of possible In-Path-Equipment (IPE), such as echo cancellers, and commands them to go inactive in TFO (if they are upgraded for the TFO protocol).

If the Codec_Types or Codec_Attributes do not match, then further parameters, like the alternative Codec_List are exchanged, if made available by the network to the TRAU.
For each system the Codec_Types are rank-ordered by implicit rules. After exchange of these Codec_Lists in both directions, both TFO_Partners apply identical implicit rules to determine the common Codec_List and from that the common Codec_Type with highest priority. Since the TRAUs themselves are not able to change the Codec_Types, they report to their controlling network entities and wait for an intra call modification. After some short while (during which the call remains in “normal tandem” connection) the Codec_Types are harmonised and TFO is possible.

2.2 TFO for the AMR

Modern Codecs, like the Adaptive Multi-Rate (AMR) Codec algorithm, however, provide some special opportunities. The AMR is currently foreseen in several slightly different forms (Codec_Types) in several systems:

- as GSM_FR_AMR (4 of 8 Modes, DTX, 40ms rate control)
- as GSM_HR_AMR (4 of 6 Modes, DTX, 40ms rate control)
- as UMTS_AMR (8 Modes, SCR, 20ms rate control)
- as TDMA_EFR (one mode: 7.40 kBit/s, UL_DTX, fixed mode)
- as TDMA_HR (4 of 6 Modes, UL_DTX, 40ms rate control)
- as PDC_EFR (one mode: 6.70 kBit/s, DTX?, fixed mode)
- in VoIP (8 Modes, DTX, 20ms rate control),
- …

It is in principle possible to establish a TFO connection between all these different AMR Codec_Types and systems, since the speech parameters are fully compatible, provided the “frame work” is also made compatible:

- Codec Mode Adaptation (in GSM) respectively Rate Control (in UMTS)
- Discontinuous Transmission (in GSM) respectively Source Controlled Rate (UMTS)
- Bad Frame Handling (Error Concealment, in all systems).

Note: The Codec_Type “GSM_EFR” is a specific mutant of the AMR algorithm. Its speech parameters are also fully compatible to the AMR mode with 12.2 kBit/s, but the DTX schemes are not compatible. Therefore a TFO connection could be established only, if DTX is disabled in both directions. Since DTX is relatively important, especially in uplink direction to save battery life, this is currently not the recommended way. Instead it is proposed that every node (TRAU and Mobile) that is capable of AMR should also implement the GSM_EFR DTX scheme (a relative small complexity increase) and thus TFO between AMR and GSM_EFR is not needed.

Back to the AMR: The AMR Codec family consists of 8 different source Codec_Modes, ranging from 4.75 kBit/s up to 12.2 kBit/s. By observing the transport channel capacity, especially that of the radio links, it is possible to adapt the source bit rate to the channel capacity for optimal throughput and speech quality.

The GSM network, represented by the serving BTS, is fully authorised to select the Codec_Mode in uplink and downlink. The MS and TRAU must fully obey the Codec_Mode_Command from the BTS.

A slightly different approach was taken for UMTS: the network, represented by the RNC, specifies the highest allowed Codec_Mode for the TC in downlink and the UE in uplink and typically the UE will use this highest mode for optimal speech quality. But the UE may select a lower Codec_Mode as well, e.g. in case it hits the upper transmit power limit. This difference to GSM needs further study in TFO connections between GSM and UMTS. It seems reasonable consider the more general UMTS approach.

A further difference between GSM and UMTS lays currently in the allowed Codec_Mode update rate: In GSM the Codec_Mode can only change in a fixed grid of 40ms (two speech frames). The TDMA structure of the BTS defines in which frames (odd or even) a change can happen. Both TFO partners need to synchronise and then stay within the once selected grid. In UMTS – according to the current specification – the Codec Mode can change every speech frame (20ms).
TFO between GSM and UMTS will not be possible without loss of speech frames in the direction from UMTS to GSM, if the UMTS side changes its Codec_Mode in the "wrong" phase for the GSM receiver. One possible way is to restrict the change of the Codec_Mode also in UMTS (in uplink direction) to the 40ms grid. It does not matter which phase is selected, but it must be kept thereafter.

And a third difference: GSM specifies that (typically) the Codec_Mode shall only be changed to its neighbouring mode (seldom exceptions allowed). The receiver can take advantage of this statistical property and achieve a lower frame error rate, especially in GSM_HR_AMR. In Codec_Type UMTS_AMR there exists – currently – no such restriction.
It seems reasonable to apply the same restriction as in GSM also in UMTS.

In a GSM environment not all eight modes are necessary during one speech call: a subset (ACS: Active Codec Set) of up to four Codec Modes is fully sufficient. Which ACS is selected depends on hardware conditions and operator´s choice and not the last on the national conditions: the ACS for Europe may look different compared to the one in USA or Asia.

At TFO setup the TFO_Partners therefore need to indicate which ACS they are currently using. TFO can only be established, when these ACSs harmonise. At call setup it would be not critical to take some more time to harmonise the ACSs on both sides by in call modification and then establish TFO. This change of the ACS must be known in all involved call nodes, especially in the MS, BTS and TRAU, respectively UE, RNC and TC.

Other reason for a subset of the full AMR set may come from network constraints, e.g. not full implementation or a resource shortage. As said: it is not very critical to harmonise the ACSs on both sides at call setup. But it may be critical or even not tolerable (due to loss in speech quality), when the ACS would change too often during the call, e.g. due to different resources before and after handover. It seems therefore reasonable to configure a network in a way that changes of the Codec_Type or Codec_Attributes (i.e. the ACS) during the call after TFO Setup are minimised or even fully avoided.

In case of an established TFO connection every change of the ACS on one side (e.g. due to handover) requires a synchronised change of the ACS also on the distant side (although there was no obvious reason for a change). With other words: the problems of one access network are exported to the other side – and that can in general be even a different system (GSM to

UMTS or vice versa). This in turn requires a quite complex and traffic synchronised in call modification across systems. It seems to be very recommendable to avoid or minimise such changes of the ACS. TFO should be disabled, if the access network is not homogenous enough.

It seems reasonable to assume that within one operator´s network the configuration can be homogenous in large regions and in call modifications can be kept on a low level.
The optimal and recommended network configuration is given, when all network elements support the full set of the Codec_Type and follow identical procedures.

The following Codec_Attributes and Parameters are exchanged by the TFO Protocol for the AMR:

- System_Identification (SysID: GSM, UMTS, TDMA, …)
- Codec_Type (CoID: GSM_FR_AMR, GSM_HR_AMR, …)
- Active_Codec_Set (ACS: 1 up to 8 Codec_Modes)
- AMR TFO Version Number (for future extensions)
- Supported_Codec_Set (SCS) and Max Number (MACS) in ACS (if only a subset is implemented)
- ACS_Optimisation_Mode (Fast by RATSCCH, or slow by out-of-band signalling)
- alternative Codec_List

It is not negotiated whether DTX is supported or not: since DTX in uplink is for all mobiles an essential power saving feature it can not be disabled. Therefore the following rule is adopted: the sender (the Mobile in uplink) can use DTX without restriction. The receiving end must in all cases be able to handle the incoming signal. If DTX in (the second) downlink is not supported, then the receiver must fill the speech pauses by a suitable comfort noise signal at either TRAU, BTS or MS side. The radio link is then kept active permanently by sending either coded comfort noise parameters or fill frames (SID_Filler) in speech pauses.

2.3 Inband Signalling when TFO is established

After the call has been setup in TFO the inband signalling takes care about:

- Frame and Phase synchronisation (in case of AMR the Codec Change phase)
- Initialisation of the TRAUs
- DTX handling
- Bad Frame Handling
- Potentially Phase Alignment (in case of AMR)
- Rate Control (in case of AMR)
- Pre-Handover Warning
- Potentially ACS subset (re-)definition (in case of AMR)

2.4 TFO Handling during Handover

As long as during handover neither the Codec_Type nor the Codec_Attributes needs to be changed the handover is comparably simple. TFO is not or only shortly interrupted and continues more or less seamless after handover. The potential TFO interruption effect is masked by the handover effects.

More complex is the case when the Codec_Type needs to be changed, e.g. because the new cell can not support the previously used Codec_Type. TFO is then not longer possible after handover. But typically the TRAUs are not pre-warned and therefore the Codec_Type mismatch comes by surprise and fast fall back to normal tandem operation is necessary.

Depending on the round trip delay between the TRAUs and as long as that there could be a audible distortion by the inserted, but unuseful speech parameters, until the call is fully back in normal mode. For that reasons the AMR standard in GSM specifies the optional “Pre-Handover Warning” from BSC to serving BTS. TFO can by that be disabled in a controlled way before the handover to keep the distortion minimal.

Possibly TFO can be renegotiated after handover and if the distant TFO Partner is able and willing to modify its radio leg as well (by out-of-band signalling) the TFO connection can re-establish with the new Codec_Type. Lesson: A change at one radio leg may cause a change at the other radio leg, too. Therefore: These changes should be kept at a minimum, both for better quality and for lower signalling load.

In case of AMR it was for a long time the working assumption that the Active Codec Set could change at every handover due to different versions of old and new BTS equipment. Since every change of the ACS requires a synchronised change at the distant side as well, and since the assumption is that the procedures should work reasonably well for a handover rate of 1/10s, it seemed unavoidable to develop the RATSCCH (Robust AMR Traffic Synchronised Control Channel) to allow these ACS modifications end-to-end without undue interruption of the speech path. RATSCCH is optional and therefore it is indicated in the TFO setup protocol whether RATSCCH is supported or not. RATSCCH can be used in an optimal way only if both sides support it.
What seemed reasonable for TFO within GSM needs reconsideration, when it comes to TFO between GSM and other systems (UMTS, …).

Recently Ericsson for these (and other) reasons proposed to reconsider the working assumptions and basically allow TFO only when the radio access network is homogenous enough so that too frequent changes of the ACS can be avoided. Ideally all network elements (MS, BTS and TRAU) should support the full AMR set, then both the TFO handling is simplest and speech quality is optimal.

3. General Principle of Transcoder Free Operation

Transcoder Free Operation has the same objectives as TFO, but it does it on a higher protocol layer by out-of-band signalling. The most important difference between TFO and TrFO is: the whole transport network control layer must be aware that coded speech is transported. Invoking of supplementary services can now only be done keeping this in mind. Typically so called “Media Gateways” (MG) are responsible to take care that the correct coding is applied. The big advantage of TrFO is: the coded traffic stream is not interrupted without careful consideration of the consequences. Transitions can be controlled in a better way.

3.1. Call Setup in TrFO by Out-of-Band Signalling

Figure 2 gives a principle of the architecture for UE-to-UE calls.

Call setup is triggered by the originating UE. It sends its capabilities (List of alternative Codec_Types) to its serving MSC. This originating MSC sends the – possibly modified – Codec_List using the BICC (Bearer Independent Call Control) along the routing path up to the other MSC controlling the terminating UE. All network nodes along the path, especially the Media Gateways (respectively its controlling MSCs) have the ability to check the Codec_List and potentially “puncture out” Codec_Types that are not or not fully supported. We know: at least G.711 will remain within the Codec_List, if nothing else.
The terminating MSC interrogates the terminating UE and finally selects the preferred Codec_Type and sends it back all the way. This is a different approach compared to TFO, where both sides have identical rights and behaviour.

One substantial difference is also that the priorities of the Codec_Types within the Codec_List in BICC can be set randomly by the originating side, but the terminating side can select in another priority. The priority in TFO, however, is predefined in the standard and can not be changed dynamically. Interoperation between TrFO and TFO may need to take this into account, otherwise not-compatible Codec_Types may be selected.

If this Codec_Type is different from G.711 then a Transcoder Free Operation can be established end-to-end. Highest possible speech quality and lowest possible transport capacity are achieved.

The UEs and the MGs are informed about the selected Codec_Type and its Codec_Attributes. In Figure 2 the H.248 protocol is assumed for the control of the MG and therefor for the transport of Codec_Type and Codec_Attibutes. At a later point we will see that this seems to be the protocol of choice for the other TFO relevant information: the Codec_List, etc.

The RNCs are not aware which Codec_Type is used: the UTRAN is configured to the application, but it does not know details of the bit stream is handles. This is to allow fast introduction of new applications into UMTS without redesigning the UTRAN.

It seems reasonable to assume that in most part of the conversation no supplementary services are involved and thus the MGs provide a transparent path between the RNCs. Whether or not the transport bearers and the associated framing protocols along the path are identical or different does not matter, as long as they are able to carry the same “logical information”. Thus virtually the RNCs talk directly to each other for most of the time. But one important prerequisite is: the CN must provide this logically transparent bearer for the full IU UP functionality.

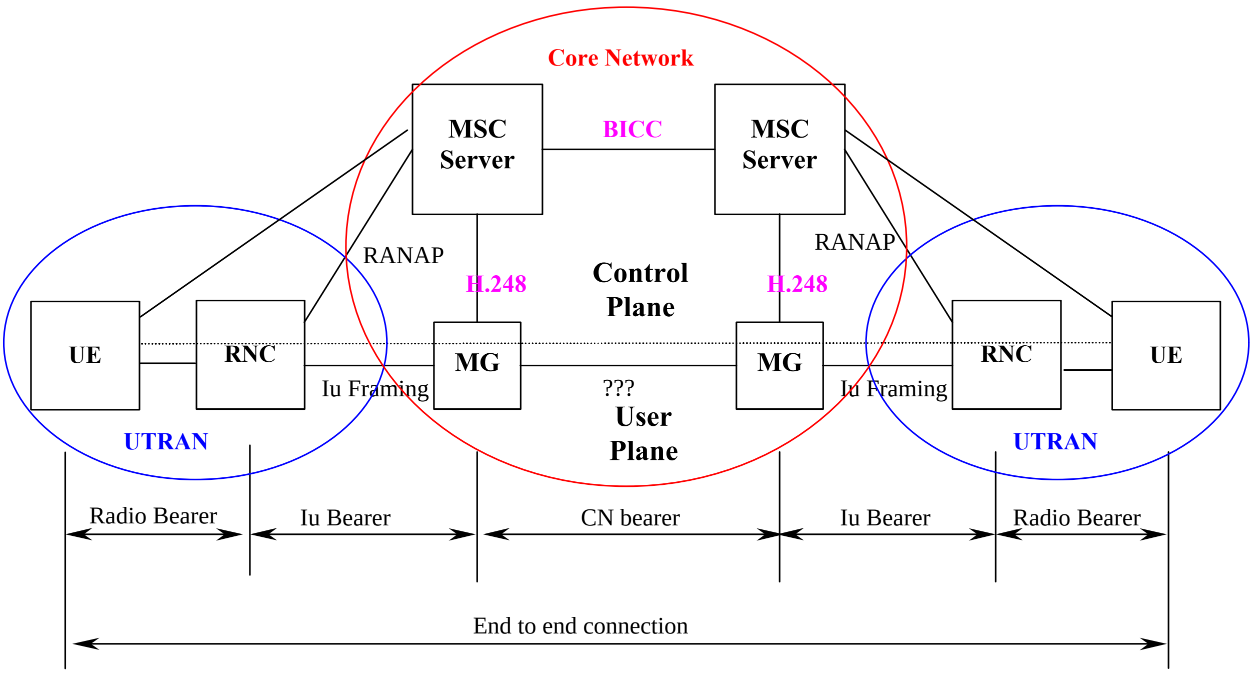

Figure 2: Functional Entities for Handling of Transcoder Free Operation in UMTS

As in case of TFO a TrFO connection can only be established, if Codec_Type and Codec_Attributes match exactly at both ends and in addition in all involved nodes along the path. It is important that the Originating MSC sends within the Codec_List only Codec_Types with Codec_Attributes it really is able to support. The same holds for the terminating MSC and all nodes in the path. Otherwise it might happen that the finally selected configurations do not match and then no speech connection is possible. Please note: there is no automatic fall back like in TFO, since no TC is allocated. This is a important difference to TFO.

If the BICC protocol did not get a supported Codec_Type back (other than G.711), then it has to allocate a TC. We could say: the TrFO part of the link is terminated within this TC.
The location of this TC can be very different.
The simplest scenario is given, when the TC sits close to the UTRAN. The CN then transports G.711 speech as today in 2G networks.
Another important scenario is given, when the TC sits at the interconnect point to the PSTN. Then transcoding to “legacy” networks takes place at the latest possible point and cost efficient transmission is achieved within the CN.

The following Codec_Attributes are exchanged by the BICC Protocol for the AMR:

- Codec_List
- Organisation_Identification (OID: 3GPP, …)
- Codec_Type (CoID: GSM_FR_AMR, GSM_HR_AMR, UMTS_AMR…)
- Active_Codec_Set (ACS: 1 up to 8 Codec_Modes)
- Supported_Codec_Set (SCS) and Max Number (MACS) in ACS (if only a subset is implemented).

Not exchanged parameters – compared to TFO – are:

- AMR TFO Version Number (for future extensions)
- ACS_Optimisation_Mode (Fast by RATSCCH, or slow by out-of-band signalling)

It needs to be seen whether these parameters are needed in TFO/TrFO and in which network node.

DTX/SCR is handled similar to the TFO approach: the sender (UE in uplink) can use DTX/SCR if appropriate, the receiver must be able to handle that always.

3.2 Inband Signalling in established TrFO

After the call has been setup and the bearers are established end-to-end the inband signalling takes care about:

- Frame/packet synchronisation
- Initialisation of the Radio Bearers
- Potentially Time Alignment (in UE-to-PSTN calls)
- SCR/DTX handling
- Bad Frame Handling
- Potentially ACS subset definition (in case of AMR)
- Rate Control (in case of AMR)

This inband signalling is very similar, if not identical to the one in case of TFO.

An interesting difference lays in the Rate Control signalling between GSM and UMTS.
In GSM the Codec_Mode_Request is sent every 40ms, regardless whether it has changed or not. This gives a high implicit robustness and transmission errors are quickly corrected.
In UMTS Rate Control commands are only sent when the (maximum) rate has to be changed. No automatic repetition is forseen. A loss of the rate Control command has an other impact than in GSM, especially it takes longer to correct the error. There is no difference between TrFO and TFO within UMTS in this respect.

4. Interaction between TFO and TrFO

Figure 3 shows a typical scenario in an UE-to-MS call (UMTS to GSM). Similar scenarios hold for all other 3G to 2G calls and 3G to 3G calls.

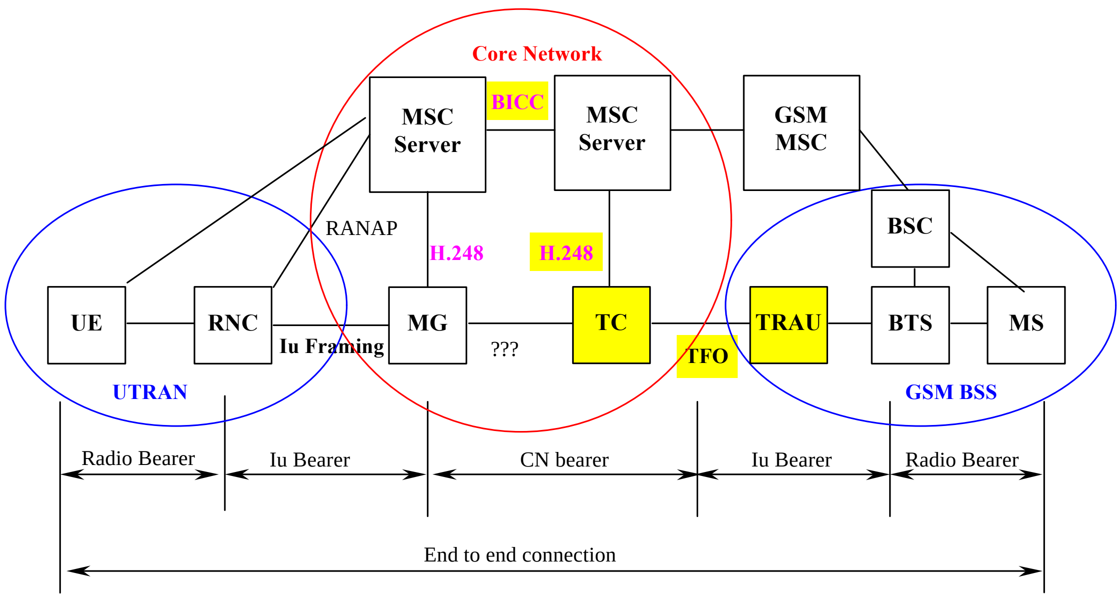

Figure 3: TrFO – TFO Call between UMTS and GSM

At call setup time it is not known, whether TrFO-TFO would be possible or not. So both systems allocate a TC respectively a TRAU to handle the speech compression for their radio access. The TC at the edge of the Core Network communicates with the TRAU within the GSM BSS via the TFO protocol. No other signalling layer exists to exchange the Codec_Type, Codec_Attributes and the Codec_List.

According to todays working assumptions the UMTS would select the UMTS_AMR Codec_Type, optimally with all 8 Codec_Modes and SCR enabled. The cell and network load will during call time constrain the highest Codec_Mode, but typically this will be the 12.2 kBit/s mode at call setup.
Note: during the last S4 meeting a LS was exchanged with other 3GPP groups stating that the Initial Codec Mode (ICM, i.e. the mode the radio link starts with) needs not be negotiated within the BICC protocol and down to the UE.

The GSM would potentially select the GSM_FR_AMR of GSM_HR_AMR – or any other Codec_Type - but not the UMTS_AMR.

Therefor an immediate TFO setup would not be possible, mainly because the Rate Control interval is incompatible (20ms in UMTS_AMR, 40ms in GSM_xx_AMR).

Two ways could improve that situation:

- per definition the UMTS_AMR should use the same 40ms Rate Control Interval,
- per definition it should be mandatory for any UE to support GSM_FR_AMR and GSM_HR_AMR – and all other Codec_Types of the AMR family.

Then the TFO_Partners (TC and TRAU) exchange the alternative Codec_Lists and potentially find common Codec_Types. But since it is today not mandatory for any UE to support more than UMTS_AMR, the likelyhood for matching Codec_Types is small, see above.

The TRAU gets the necessary TFO parameters from the BSC via the BTS by inband signalling (Configuration Frames).
From where does the TC get the TFO parameters? It seems most reasonable that the MSC server in charge of the MG control provides the TFO Parameters via the H.248 protocol. This is underlined by the fact that this server has the information already via the BICC protocol. The H.248 today does not support all of these parameters, especially not the Codec_List.

In case of Codec_Type mismatch and after exchange of the Codec_Lists the Core Network needs to be informed about the distant TFO Configuration. The most reasonable way seems to be the H.248 and the BICC to transmit the distant TFO Configuration.

Both sides consider now both Codec_Lists and try to determine the common Codec_Type. It is of high importance that both sides come to the same selection.

Last remark: TFO between two UMTS networks will (at least at the startup of UMTS) always select the same mandatory UMTS_AMR Codec_Type. It seems reasonable that they both support all 8 AMR modes including SCR, because then immediate TFO setup is always possible.

5. Conclusion

The presented list of open issues may not be comprehensive and other points may come up. In any case a careful gross-check of relevant call scenarios is necessary to verify that in all cases TrFO-TFO interaction leads to reasonable call configurations.

Guidelines for the operators how to configure the networks and when to enable TFO may be helpful.

All involved 3GPP working groups need to work together to perform this tasks.