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This is an updated (v 2.0.0) stage 2 specification on out of band transcoder control from Milan meeting and following e-mail discussion.

It is proposed that the OoBTC stage 2 (TS 23.153) is to submit to CN#7 plenary meeting for information and approval.

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Technical Specification

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Foreword

This Technical Specification has been produced by the 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 this TS, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version 3.y.z

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x the first digit:

1 presented to TSG for information;

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3 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 specification.

1 Scope

This Technical Specification specifies the stage 2 description of the Out-of-Band Transcoder Control.

Cellular networks depend heavily on codecs to provide their services. Codecs are necessary to compress speech in order to utilise efficiently the expensive bandwidth resources both in the radio interface and in the transmission networks.

Transcoding of speech significantly degrades quality and, therefore, cellular systems try to avoid it for mobile-to-mobile calls when both UEs and the network support a common codec.

Digital cellular systems support an increasing number of codec types. As a result, in order to allocate transcoders for a call inside the network, and to select the appropriate codec inside the UEs, signalling procedures are defined to convey the codec selected for a call to all the affected nodes (UEs and transcoding points inside the network). Also, codec negotiation capabilities are being defined to enable the selection of a codec supported in all the affected nodes, i.e. to resolve codec mismatch situations. This codec negotiation maximizes the chances of operating in compressed mode end-to-end for mobile-to-mobile calls.

Although the main reason for avoiding transcoding in mobile-to-mobile calls has been speech quality, the transmission of compressed information in the CN and CN-CN interface of the cellular network also offers the possibility of bandwidth savings.

To also allow transport of information in a compressed way in transmission networks, these networks make use of the Bearer-independent Call Control (BICC) protocol as BICC provides means for signalling codec information.

Out-of-Band Transcoder Control bases on the possiblilities the BICC protocol offers for the negotiation and selection of codecs end-to-end.

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.

A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[1] 3G TS 23.107: "QoS Concept and Architecture"

[2] 3G TS 24.008: "Mobile radio interface layer 3 specification Core Network Protocols –Stage 3"

[3] 3G TS 25.413: "UTRAN Iu Interface RANAP Signalling"

[4] 3G TS 25.415: "UTRAN Iu Interface User Plane Protocols"

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

[6] Q.1901: "Bearer Independent Call Control"

[7] Q.765.5:"Application Transport Mechanism for Bearer Independent Call Control"

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this specification the following definition apply:

BICC: The BICC protocol includes the following two capabilities:

a. Capability to convey codec information to all the nodes with coding/transcoding functionality (terminals/access nodes and network nodes).

b. Capability to negotiate, among all the BICC nodes with coding/transcoding functionality, a common codec to be used for a specific call.

These two capabilities must be supported prior to the commitment of bearer resources to allow the optimal bearer resources to be allocated. These capabilities enable seamless inter-working/ convergence between mobile and fixed networks using BICC when compressed speech is deployed in both scenarios.

Some limitations applied for Release 99. See subclause 4.4 (BICC CS1).

Codec: A codec is a device to encode information from its original representation into an encoded form and to decode encoded information into its original representation.

Tandem Free Operation: Tandem Free Operation is the configuration of a connection with two transcoders that support TFO protocol and whose external coding schemes are compatible, thus enabling compressed speech to pass between them. When the TFO protocol is not supported by both transcoders or the coding schemes are not compatible then normal "Tandem" operation occurs and PCM encoded speech is passed between them.

Transcoder: A transcoder is a device to change the encoding of information from one particular encoding scheme to a different one. Most commonly to/from a compressed speech algorithm from/to PCM.

Transcoder Free Operation: Configuration of a speech or multimedia call for which no transcoder device is physically present in the communication path and hence no control or conversion or other functions can be associated with it.

Out of Band Transcoder Control: Capability of a system to negotiate the types of codecs and codec modes on a call per call basis through out-of-band signalling. Out-of-Band Transcoder Control is required to establish Transcoder Free Operation.

3.2 Abbreviations

Abbreviations used in this specification are listed in GSM 01.04.

For the purposes of this specification the following abbreviations apply:

APM	Application Transport Mechanism
BC	Bearer Control
BICC	Bearer Independent Call Control
CC	Call Control
OoBTC	Out-of-Band Transcoder Control
QoS	Quality of Service
RAB	Radio Access Bearer
TFO	Tandem Free Operation
TrFO	Transcoder Free Operation
UP	User Plane

4 Out-of-Band Transcoder control functionality

4.1 OoBTC Requirements

The OoBTC mechanism shall support the following:

- The capability to negotiate the preferred codec type to be used between two end nodes and to avoid the use of transcoders in the network.
  - The originating UE indicates the list of supported codecs for codec negotiation. This list shall be conveyed to the terminating MSC. The terminating MSC shall compare this list with those supported by the terminating UE and shall endeavour to select a compatible codec type. Where more than one compatible codec type is available the highest in the common list from the originating UE and terminating UE shall be selected.

    Note: For a codec type supporting various modes, the described functionality shall also be applicable to negotiate the set of codec modes common to originating and terminating UEs. The negotiations such as Initialisation and Rate control are performed by the Iu UP protocol.
  - The terminating UE indicates its list of supported codec types to the terminating MSC.
  - Where no compatible coding scheme can be selected between the UEs then the default PCM (G.711) coding shall be selected. The originating MSC shall insert a transcoder in the path from the originating UE. Codec selection for the terminating UE is then performed within the terminating MSC, independently of the originating MSC.
- The capability to control the presence of transcoders in the network.
  - Where a change to the call state of a transcoder free connection occurs, such that compressed speech cannot be maintained, it shall be possible to insert a pair of transcoders where needed in the path and inform the end points of this segment that the speech coding is changed to PCM. Such examples where this could occur are:
    - SS interruptions (e.g. A to B call connection becomes to multiparty call connection.)
    - DTMF signal insertion.
    - Handover to an incompatible partner.
    - Synchronisation loss
  - Where a change in call state as described above is temporary then it shall be possible to return to a transcoder free connection by removing the inserted transcoders and informing the endpoints that the connection has resumed to compressed speech encoding.
- The codec types comprise codecs for speech. The transcoder control should have enough expandability to support future enhancements of codec types.
- The transcoder control procedure shall not cause a perceivable time lag in the cases of establishing transcoder free connection and reverting to normal (double transcoded) call connection. In the cases described above for control of the presence of transcoders.

4.2 Relationship between OoBTC and In-band TFO

OoBTC is used to attempt to establish a UE-UE transcoder free connection. In-band TFO protocol is activated only if transcoders in tandem (a pair of transcoders with PCM coded between them) are able to communicate to each other (both supports TFO). The inband TFO protocol then allows the transcoders to compare coding schemes and if compatible to overwrite the PCM with the pure compressed speech (effectively bypassing the transcoding). The process for in-band TFO can only be performed after the call connection has been established.

If the OoBTC fails to establish the TrFO and transcoders are required, then in-band TFO may be used after call set-up. Inband TFO shall be the fallback mechanism when transcoders cannot be avoided, either at set-up or during the communication phase. In-band TFO shall be used for interworking with the 2G system (i.e. GSM).

4.3 Lawful interception

The TrFO shall be maintained if the interception is made due to the lawful interception. Two decoders are needed to monitor the TrFO call.

4.4 OoBTC restrictions/limitations for Release 99

The following restrictions/limitations are applied for Release 99.

- The BICC Capability Set 1 is applicable to this specification. It has the following limitations:

    a. Only AAL1 & AAL2 bearers are considered.

    b. No Bearer modification is supported.

    c. No re-negotiation of the supported codec list is supported. Only rationalisation of the supported list is possible.

- The BICC protocol shall be independent from the location of the transcoder in the network. However, the transcoder has to always be located in the serving MSC in Release 99. Interactions between OoBTC and transcoder location at the edge of PLMN are to be studied for Release 00 or later.
- In some handover situations (3G MSC to 2G MSC, 3G MSC to 3G MSC with TDM link) it is possible to have a configuration with 4 transcoders on the speech path end-to-end. How to avoid such a situation has to be studied for Release 00 and later. See subclause 6.5.1 for detail.

5 Network model

5.1 General

The codec negotiation mechanism is designed to work in the general situation where more than two call control (CC) nodes need to participate in the codec negotiation. The codec negotiation mechanism works as follows:

- Initiating CC node: sends its list of supported options with the level of preference associated to each one.
- Transit CC nodes: if needed, analyse the received list of options, delete unsupported options from the list and forward the list. No modification is done to the preference levels of any of the listed codecs.
- Terminating CC node: analyse the received list of options with their associated priorities and selects the supported option with higher indicated priority.

Figure 5.1/1 illustrates the mechanism. The negotiation occurs at call set-up phase only. However, as described in the next section, it shall be possible to modify the selected codec at any moment during the active phase of the call. This figure is BICC-based, the ISUP-case will be identical but without the bearer control (BC) signalling flow.

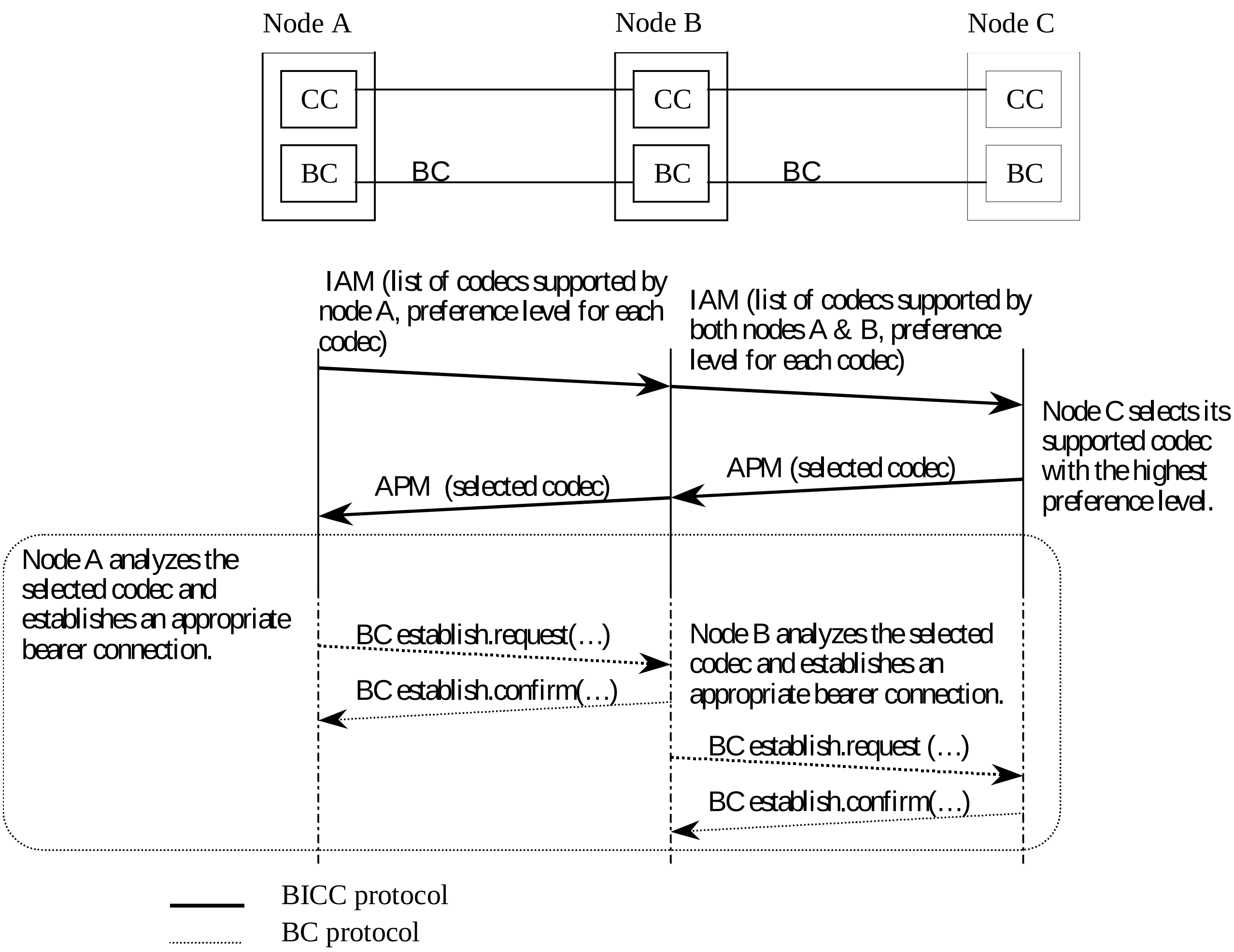


Figure 5.1/1. Sequence of BICC and BC messages for the proposed codec negotiation

The following sections describe successful call establishment scenarios using the codec negotiation mechanism.

5.2 Simple call set-up

The signalling flow for the simple call set-up case is illustrated in figure 5.1/1. Codec negotiation is done prior to the establishment of bearer connections, so that appropriate bearer resources are committed to the call. In the proposed sequence, the codec negotiation starts with the IAM message containing the list of supported codecs. The selected codec is conveyed in an APM message.

5.3 Interactions with IN and CFNRy SS at call set-up

In some cases, IN services (e.g. prompting) are triggered at CC-IN nodes that require the establishment of an user plane (UP) bearer for tones or announcements to be sent to the calling party. In order to establish this bearer, it is necessary that the CC-IN node temporarily selects one codec in the codec list sent from the initiating node, and informs the initiating node of the selected codec. Afterwards, the call may continue its establishment to the another node, which may not support the selected codec but requests that another one in the list be selected instead.

A similar situation arises with the CFNRy supplementary service. A UP connection needs to be established between the originating and “provisional” terminating CC nodes to enable ringing tones to be sent to the calling party. The type of codec must be agreed prior to the establishment of the bearer connection. Afterwards, the call is redirected to another node that may not support the selected code but requests the selection of another one.

Figure 5.3/1 shows how the proposed codec negotiation works in these two cases. A procedure for modifying the selected codec is needed to cope with these cases. For the BC level, Figure 5.3/1 gives an example that includes procedures to modify the bearer. If the BC does not have this modification capability, the bearer would have to be modified by releasing and re-establishing bearer connection.

If the node C returns the same codec type as a codec selected between node A and node B, then procedure between APM (modify to codec y) and APM (modify comfirm) shall be skipped.

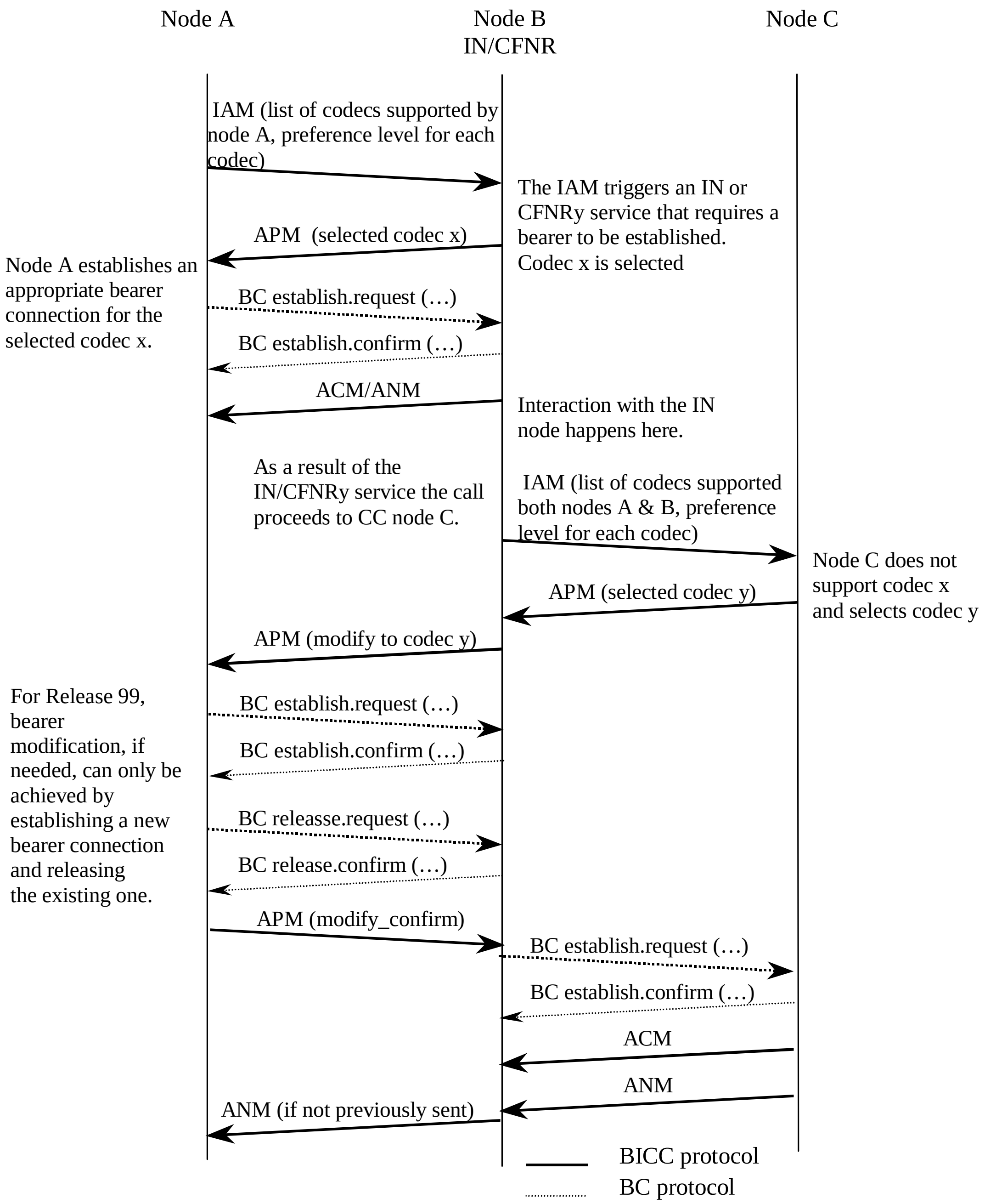


Figure 5.3/1. Sequence of BICC and BC messages for case of IN prompting services and CFNRy

5.4 Conference calls

*[Editor's note: **This section might need further updates.** RAN3 plans to find suitable solution about this open issue in their March 2000 meeting. Thereafter, The description in this subclause will be reconsidered when RAN3 has decided on in-band or out-of-band solution.]*

Conference (multi-party) call service is provided by means of a conference call device (CCD) located in a node inside the network. A call leg is established between the CCD and each party participating in the conference. Since the CCD operates only with PCM/analogue speech formats to mix the speech signal from the different conference call legs, codec negotiation procedures must be carried independently for each call leg between the node where the CCD is located and the parties at the other end of each leg. That means a transcoder is in principle[1] allocated to each conference call leg, and different call legs can actually operate in different compressed speech formats.

Figure 5.4/1 illustrates the BICC signalling sequence with codec negotiation for a three-party (A, B and C) conference call with the CCD device located in network access node A. Node A has transcoding capabilities for different codecs (x and y). Previous to establishing the conference, a connection has been setup between nodes A and B with codec x. Independent codec negotiation is carried out between nodes A and C resulting in the selection of codec y for this leg of the conference call.

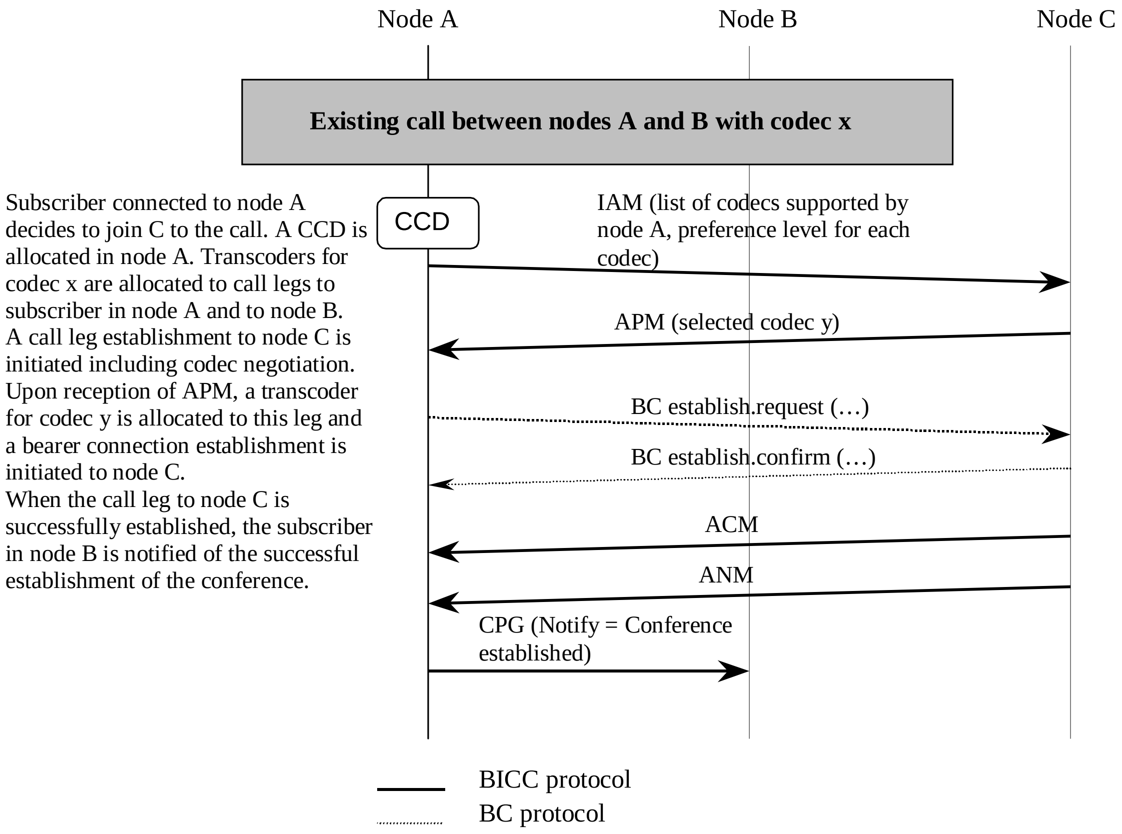


Figure 5.4/1. Sequence of BICC and BC messages for the case of conference calls.

[1] A transcoder does not need to be allocated to a call leg that selects to operate in PCM format.

5.5 Interworking with ISDN/PSTN

Refer to the subclause 6.1.2 and 6.2.2 for the interworking with ISDN/PSTN.

5.6 Adaptable network configurations for BICC in Release 99

The ITU-T has generated the following BICC related specification in December 1999 for determination.

- Q.1901 Bearer Independent Call Control
- Q.765.5 Application Transport Mechanism - BICC
- Operation of the BICC protocol with AAL type 2 CS1
- Operation of the BICC protocol with B-ISUP protocol for AAL type 1adaptation
- Operation of the BICC protocol with DSS2 (Digital Subscriber Signalling System No2)

The specification Q.1901 and Q.765.5 define the mechanism of the BICC and basic parameters. The other suppliantly specifications are generated corresponding to the bearer (AAL1, AAL2, and DSS2) that might be deployed under the BICC protocol. It means, AAL type 2 CS1 and AAL type 1 are only adaptable bearers that can be used for Out of band transcoder control in Release 99. I.e., the STM cannot be used for Out of band transcoder control.

6 Information flows

6.1 Information flows for MO calls

The following sub-sections provide information flows for successful and unsuccessful invocation of TrFO in MO call establishment scenarios.

6.1.1 Successful TrFO MO call

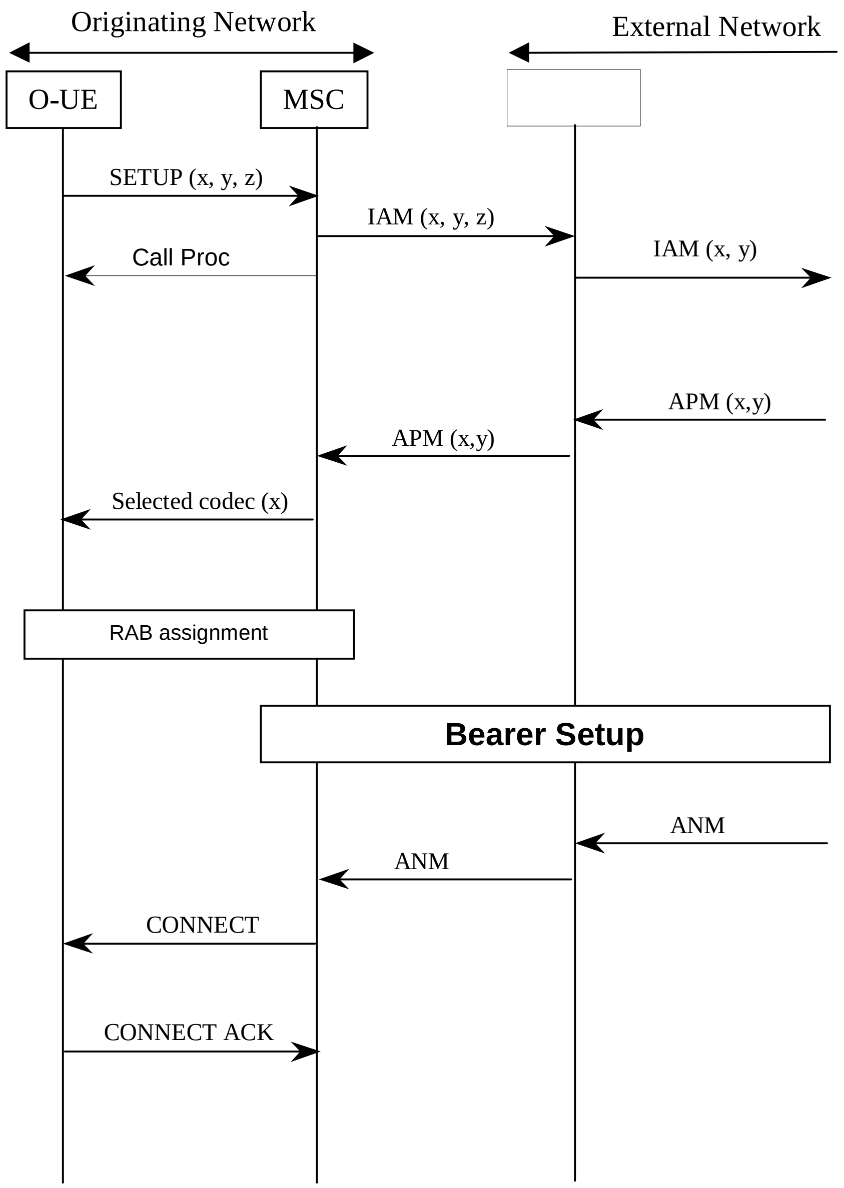


Figure 6.1.1/1 Successful TrFO MO call

- The list of codecs provided by the O-UE in the SETUP message is mapped into the codec list of the BICC protocol in the MSC. Refer to the subclause 9 for mapping rules.

- The codec list is analysed in all the BICC nodes participating in the negotiation. Non-supported codec options are removed from the list in each of these BICC nodes. For example in the figure, the codec z was screened out in the MSC.

- The codec x and y were selected as the bearer to be supported in the terminating node and this information is carried back to the MSC in an APM message. In those sections in the network using BICC, the bearer connection for the call applying codec x is established upon receipt of the APM message with codec x indicated as the highest priority. The bearer has to be set up after receiving the APM (codec x,y) message because the size of the bearer is dependent upon the codec selected.

- The O-UE is informed of the selected codec in the Selected codec message, and RAB assignment will occur.

6.1.2 No TrFO in MO call due to transit network not supporting BICC

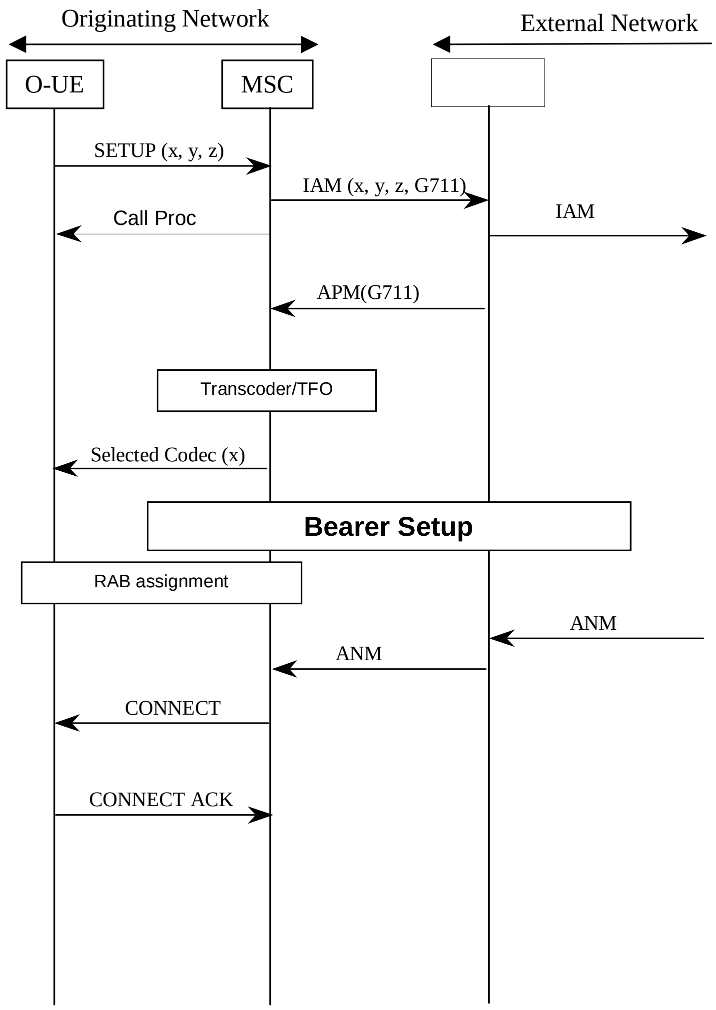


Figure 6.1.2/1 No TrFO in MO call due to transit network not supporting BICC

- The list of codecs provided by the O-UE in the SETUP message is mapped into the codec list of the BICC protocol in the MSC. MSC shall add the G711 to the codec list with the lowest priority as the preparation for BICC negotiation failure. Refer to the subclause 9 for mapping rules.
- If an intermediate node detects that the external network does not support BICC, then an intermediate node performs interworking between BICC & ISUP. This means that codec negotiation is terminated in an intermediate node. An intermediate node returns the APM with G711 codec.
- Upon reception of the APM with G711 in the MSC, the MSC determines that out-of-band codec negotiation is not possible end-to-end and a transcoder is allocated.
- In the Selected codec message, the O-UE is informed that codec x has been selected, and RAB assignment will occur.

6.1.3 No TrFO in MO call due to APM-user not supported

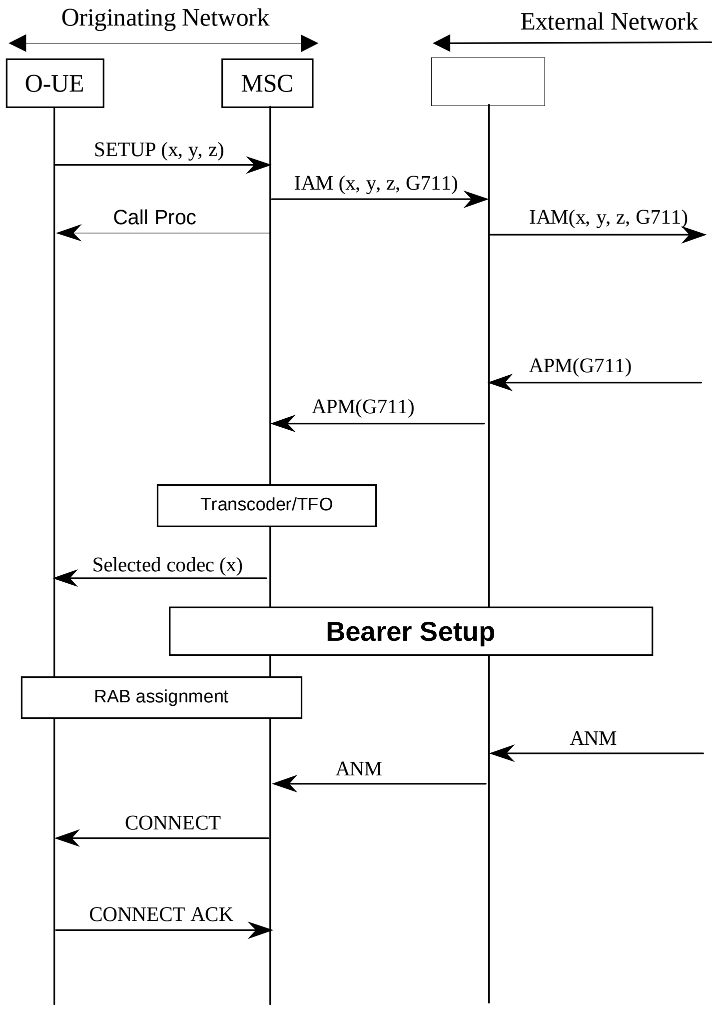


Figure 6.1.3/1 No TrFO in MO call due to APM-user not supported

- The list of codecs provided by the O-UE in the SETUP message is mapped into the codec list of the BICC protocol in the MSC. MSC shall add the G711 to the codec list with the lowest priority as the preparation for BICC negotiation failure. Refer to the subclause 9 for mapping rules.

- On reception of the APM with G711 due to some reasons. Ex.) APM-user is not supported in the terminating node. In this case, an intermediate node passes this message to MSC. Then, the MSC knows that out-of-band codec negotiation is not possible end-to-end and a transcoder is allocated.

- In the selected codec message, the O-UE is informed that codec x has been selected, and RAB assignment will occur.

6.2 Information flow for MT calls

The following sub-sections provide information flows for successful and unsuccessful invocation of TrFO in MT call establishment scenarios.

6.2.1 Successful TrFO MT call

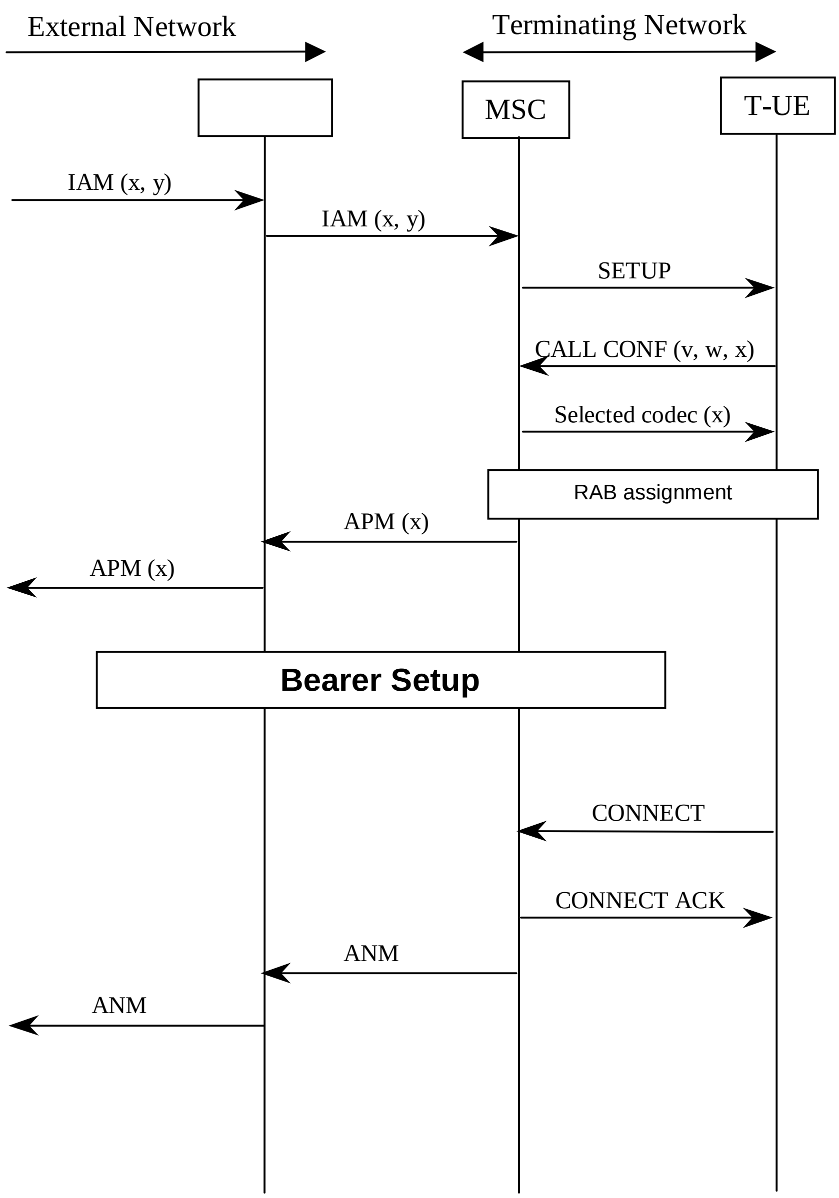


Figure 6.2.1/1 Successful TrFO MT call

- The list of codecs provided by the external network in the IAM is mapped into the codec list of the access protocol. This information is stored intermediately in the MSC and the SETUP message is sent to the T-UE. Refer to the subclause 9 for mapping rules.
- The T-UE returns its list of available codecs in the CALL CONFIRMED message.
- The MSC selects the codecs from the lists received from the T-UE and the external network and indicates this choice to the T-UE in the Selected codec message.
- The codec x selection is carried back to the originating MSC in an APM message. In those sections in the network using BICC, the bearer connection for the call with codec x is established upon receiving the APM message. The bearer has to be set up after receiving the APM (codec x) message because the size of the bearer is dependent upon the codec selected.

6.2.2 No TrFO in MT call due to APM-user not supported

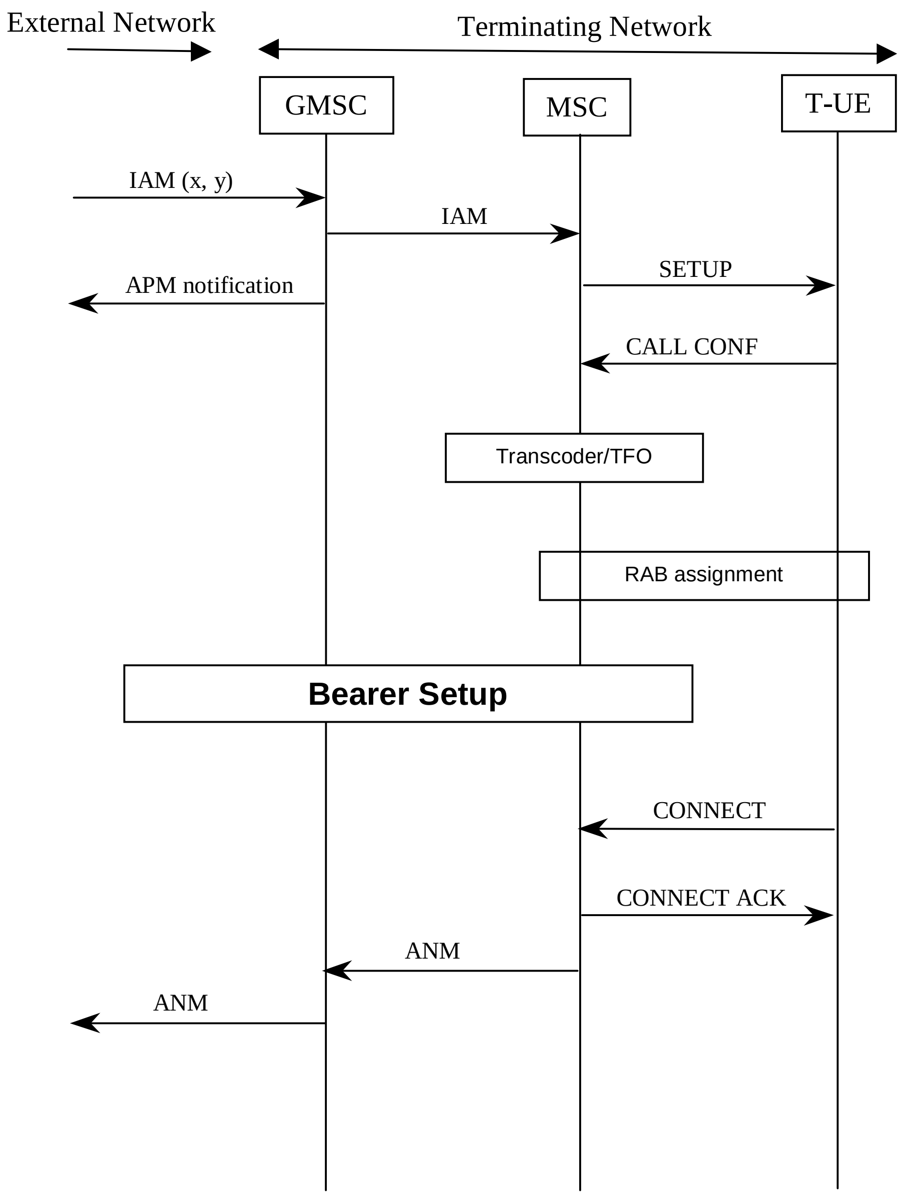


Figure 6.2.2/1 No TrFO in MT call due to APM-user not supported

- The list of codecs provided by the external network in the IAM is carried to the GMSC.
- The terminating network supports APM but not the APM-user that performs speech codec negotiation. In this case, the GMSC will follow the actions indicated in the application transport instruction indicators (ATII) of the APPs in the IAM message, which should be set to continue the connection but notify the peer. Another possibility is that the APM-user is actually supported but it is configured to not carry out negotiation (it may perform other functions as well), in this case the APM-user will implement procedures to deny such negotiation.
- The MSC determines that out-of-band codec negotiation is not possible end-to-end and a transcoder is allocated. RAB assignment will take place.

6.3 Information flow for interactions with IN and CFNR SS at call setup

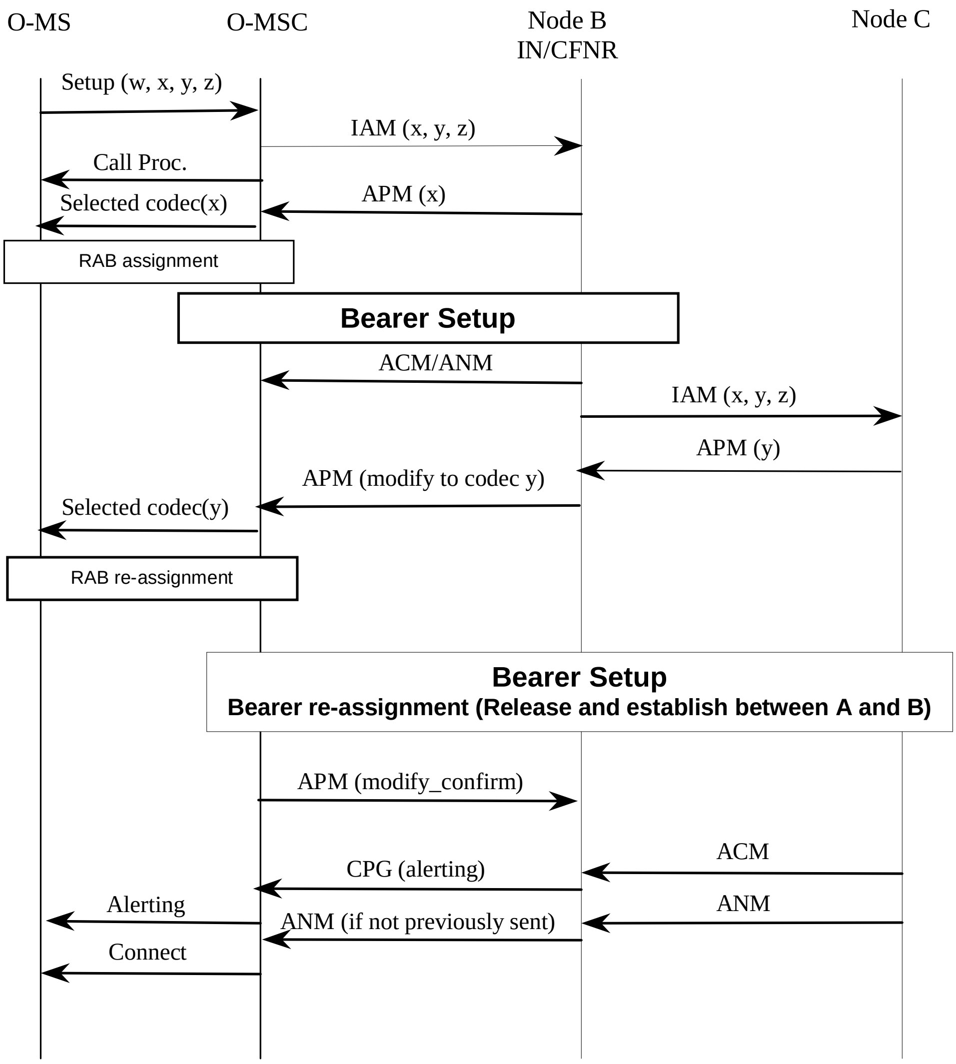


Figure 6.3/1 Interactions with IN or CFNR SS at call setup

In some cases, IN services (e.g. voice prompting) are triggered at CC-IN nodes that require the establishment of an UP bearer for tones or announcements to be sent to the calling party. In order to establish this bearer, it is necessary that the CC-IN node temporarily selects one codec from the codec list sent from the initiating node, and informs the initiating node about the selected codec. Afterwards, the call may continue its establishment to the another node, which may not support the selected codec but requests that another one in the list be selected instead.

A similar situation arises with the CFNR supplementary service. A UP connection needs to be established between the originating and “provisional” terminating CC nodes to enable ringing tones to be sent to the calling party. The type of codec must be agreed prior to the establishment of the bearer connection. Afterwards, the call is redirected to another node that may not support the selected code but requests the selection of another one.

Figure 6.3/1 shows how the proposed codec negotiation would work in these two cases. As can be seen, a procedure for modifying the selected codec is needed to cope with these cases.

If the node C returns the same codec type as a codec selected between O-MSC and node B, then procedure between APM (modify to codec y) and APM (modify comfirm) shall be skipped.

6.4 Information flow for interaction with Multiparty SS

*[Editor's note: **This section needs further updates.** RAN3 plans to find suitable solution about this open issue in their March 2000 meeting. Thereafter, The description in this subclause will be reconsidered when RAN3 has decided on in-band or out-of-band solution. The possible candidate solutions are described for the time being.]*

[Alternative 1: Using C-plane for TrFO break]

After having established a call (using codec x), the subscriber sets up another call (using codec y). When joining these calls to a multiparty call, the negotiated codecs remain active for the call leg from a subscriber to the CCD. Before the allocation of the transcoder, RFCI set report procedure is invoked. MSC A obtains the used RFCI set from RNC with the procedure. See TS 25.413 that defines it in detail.

When the procedure ends successfully, MSC A inserts the transcoders for A-party and B-party in which the permitted rate is set. Also, MSC A inserts the transcoder for C-party.

When the procedure ends unsuccessfully, the C-party addition fails.

At the CCD, the encoded speech signal is transcoded to PCM. After joining the input signals, the joint speech signal is fed back to the participants of the Multiparty call by transcoding it to the previously negotiated encoding scheme of that particular subscriber.

After drop(s) of the leaf-party and only two parties remain in the call, MSC-A may remove the transcoders, then the call may come back to TrFO condition.

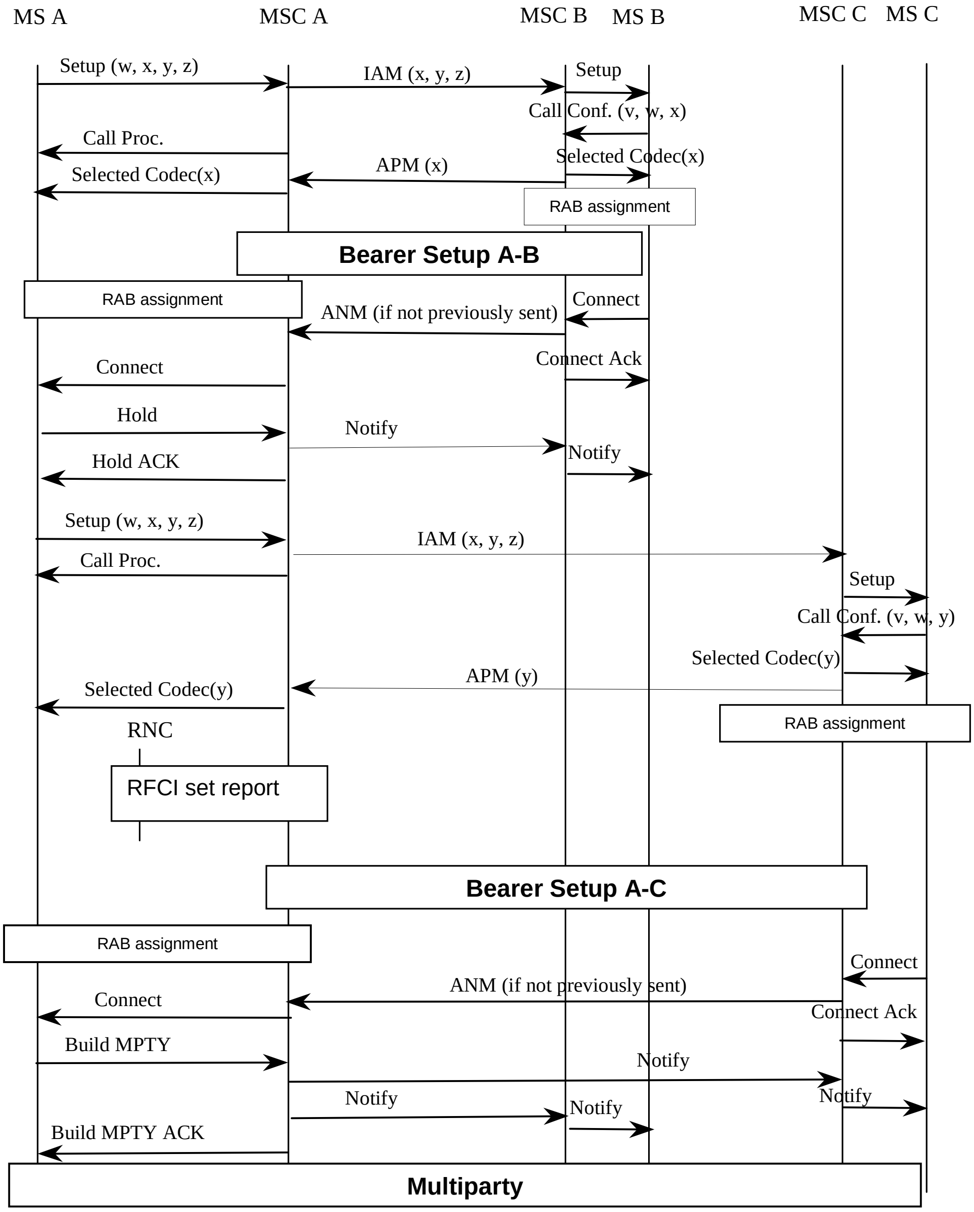


Figure 6.4/1 Interactions with Multiparty SS

[Alternative 2: Using U-plane for TrFO break]

After having established a call (using codec x), the subscriber sets up another call (using codec y). When joining these calls to a multiparty call, the negotiated codecs remain active for the call leg from a subscriber to the CCD. MSC-A inserts the transcoders for A-party and B-party. After the insertion of the transcoders, the procedure for TrFO break is invoked. See TS 25.415 that defines it in detail. The procedure doesn't interact with any call/bearer control entity in MSCs.

At the CCD, the encoded speech signal is transcoded to PCM. After joining the input signals, the joint speech signal is fed back to the participants of the Multiparty call by transcoding it to the previously negotiated encoding scheme of that particular subscriber.

After drop(s) of the leaf-party and only two parties remain in the call, MSC-A may remove the transcoders, then the call may come back to TrFO condition.

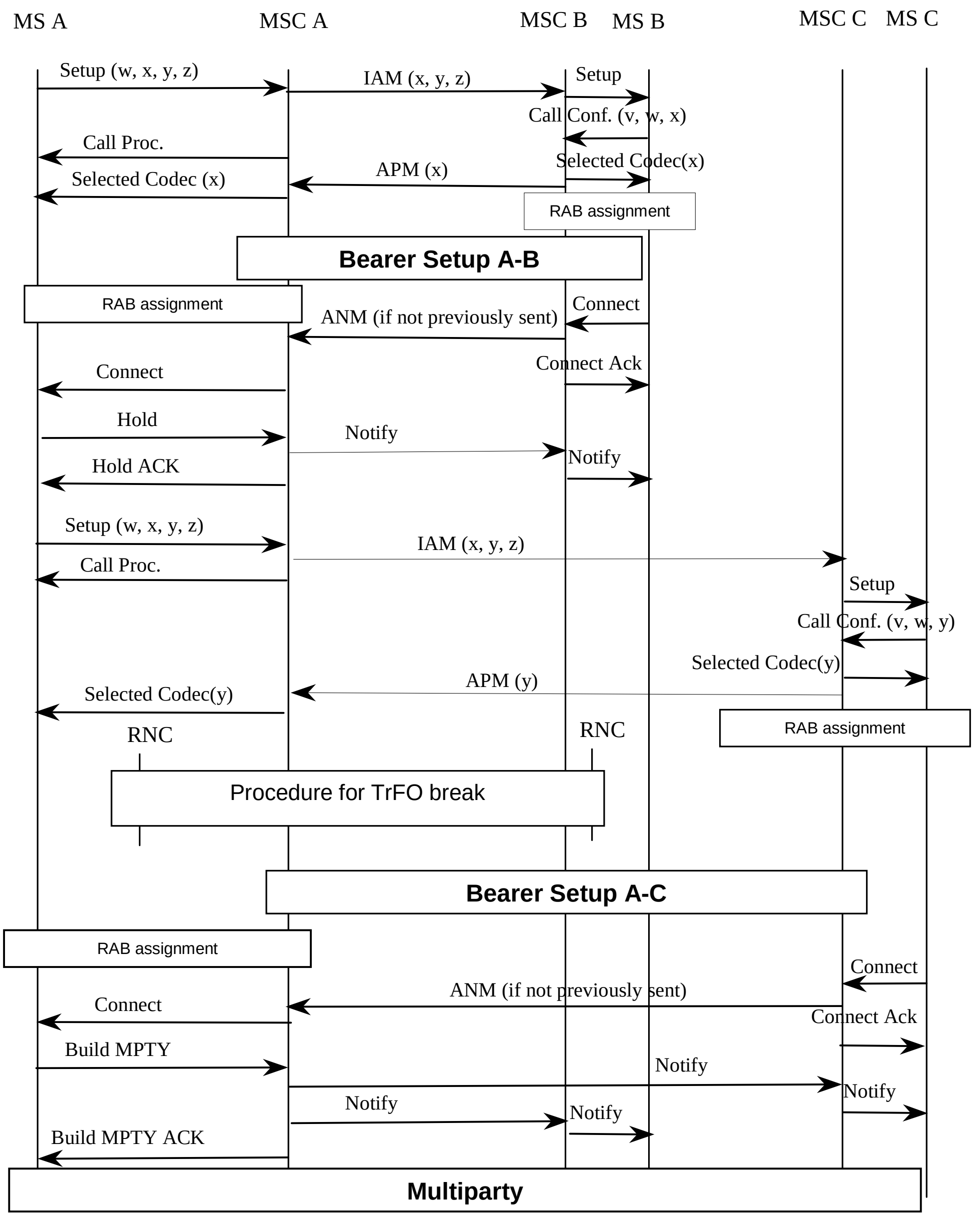


Figure 6.4/1 Interactions with Multiparty SS

6.5 Information flow for handover from UMTS to GSM after TrFO establishment

*[Editor's note: **This section needs further updates.** RAN3 plans to find suitable solution about this open issue in their March 2000 meeting. Thereafter, The description in this subclause will be reconsidered when RAN3 has decided on in-band or out-of-band solution. The possible candidate solutions are described for the time being.]*

[Alternative 1: Using C-plane for TrFO break]

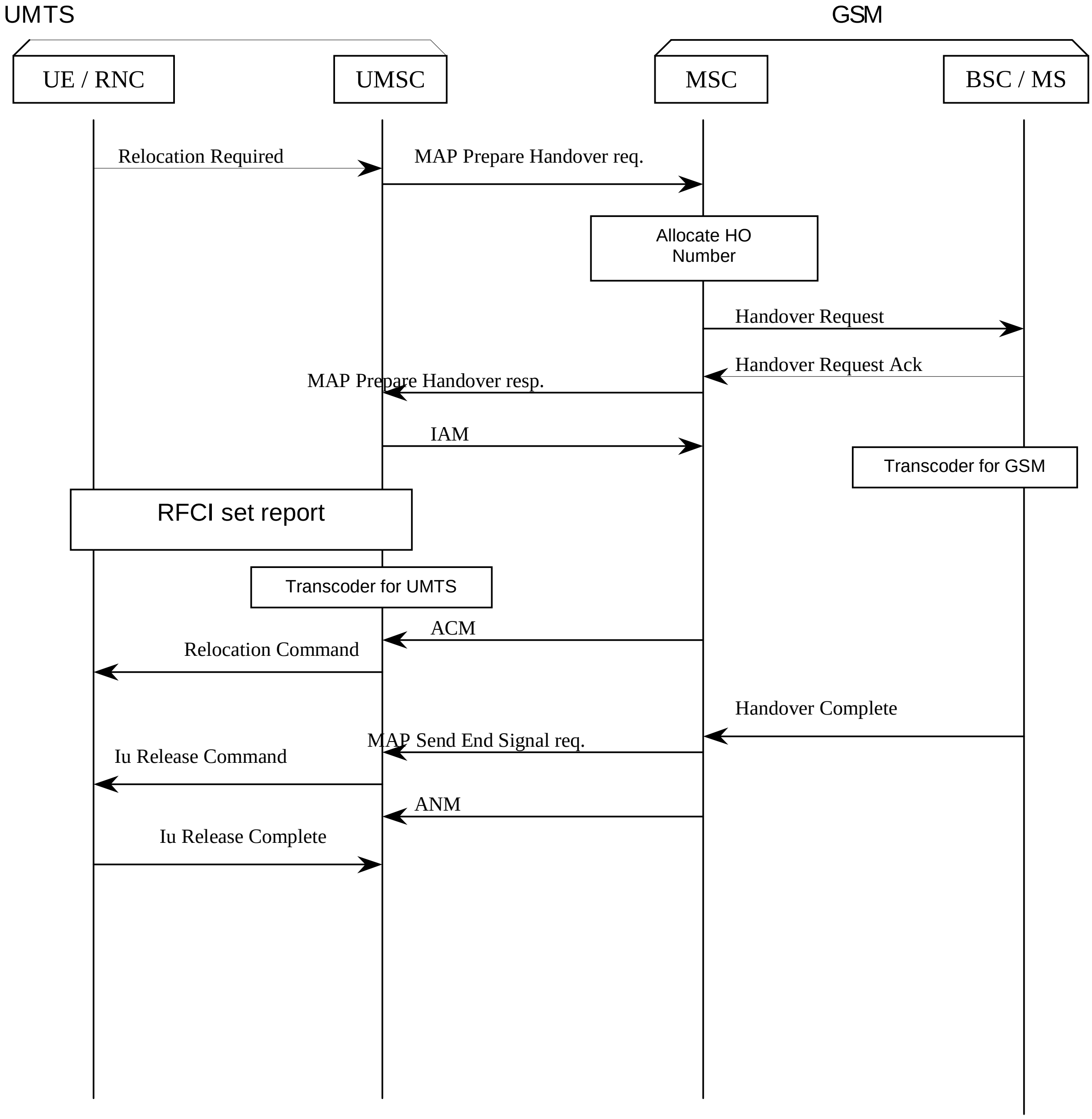


Figure 6.5/1 UMTS to GSM Handover after TrFO establishment

Figure 6.5/1 illustrates the way that transcoding will be handled for inter MSC Handover from UMTS to GSM. If the transport link between the UMSC and the MSC is TDM, then the UMSC invokes the RFCI set report procedure to obtain the used RFCI set from RNC. See TS 25.413 that defines it in detail.

When the procedure ends successfully, the UMSC inserts the transcoder for the peer UE, in which the permitted rate is set.

When the procedure ends unsuccessfully, the handover fails.

The GSM BSC will perform transcoding in the same manner, which currently used in GSM.

[Alternative 2: Using U-plane for TrFO break]

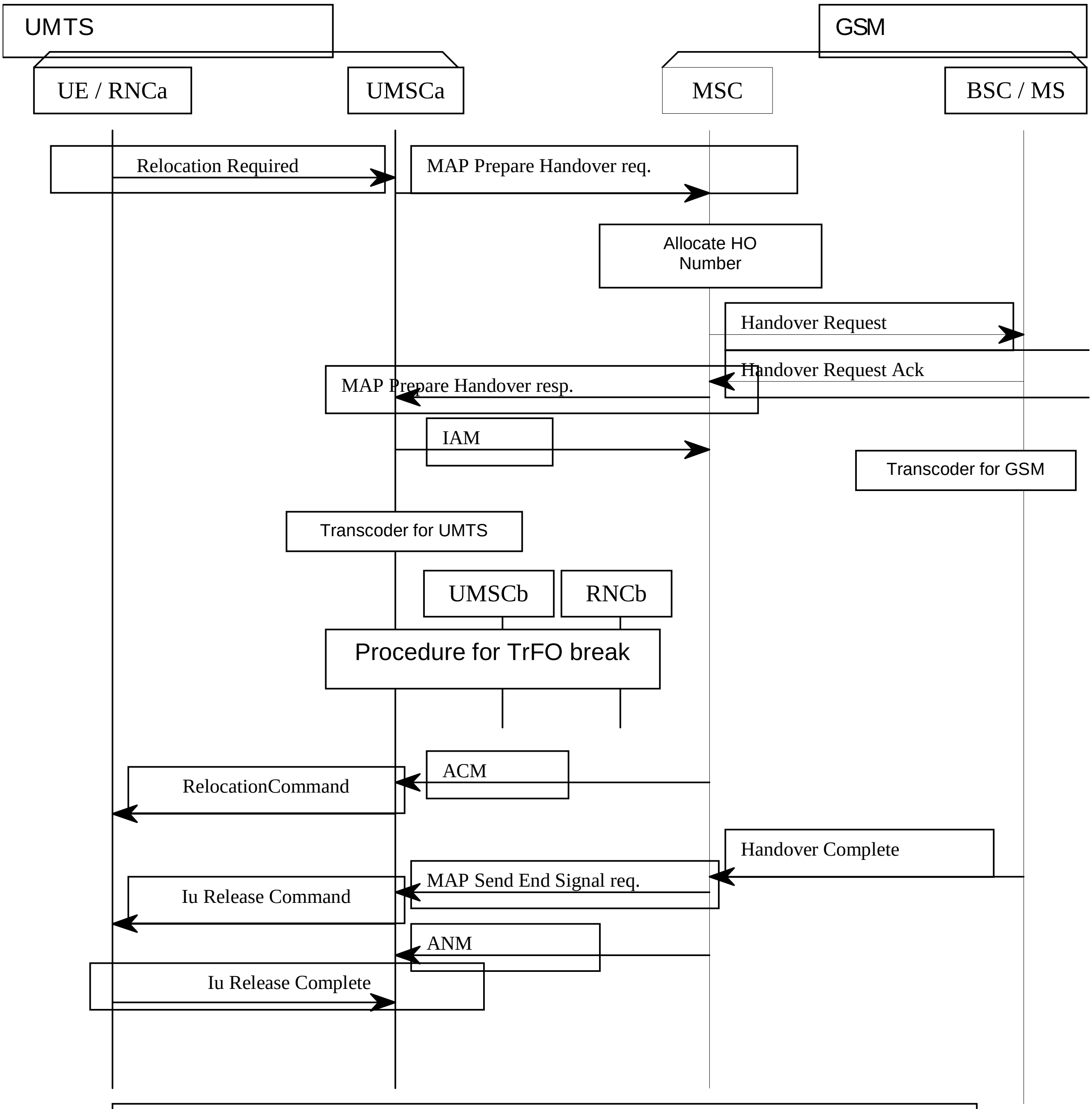


Figure 6.5/1 UMTS to GSM Handover after TrFO establishment

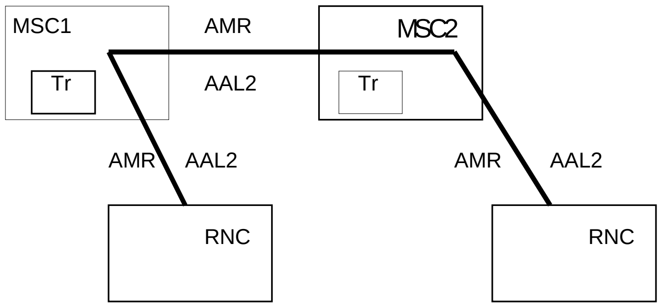
Figure 6.5/1 illustrates the way that transcoding will be handled for inter MSC Handover from UMTS to GSM. If the transport link between the UMSC and the MSC is TDM, then the UMSCa inserts the transcoder and the procedure for TrFO break is invoked between the transcoder and RNCb. See TS 25.415 that defines it in detail. The procedure doesn’t interact with any call/bearer control entity in MSCs.

The GSM BSC will perform transcoding in the same manner, which currently used in GSM.

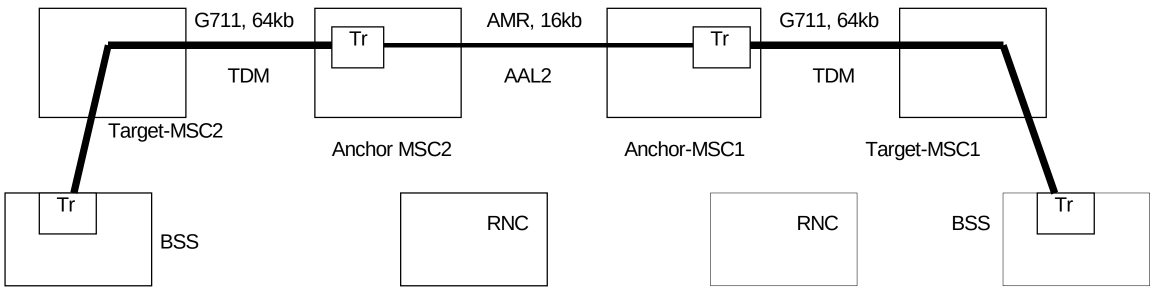
6.5.1 To be treated in future release

- 3G to 2G Handover occurs when TrFO connection in both sides.

Situation: MS-MS intra PLMN call established “Transcoder Free”, with a low rate (e.g. 16kbs) AAL2 connection for AMR .

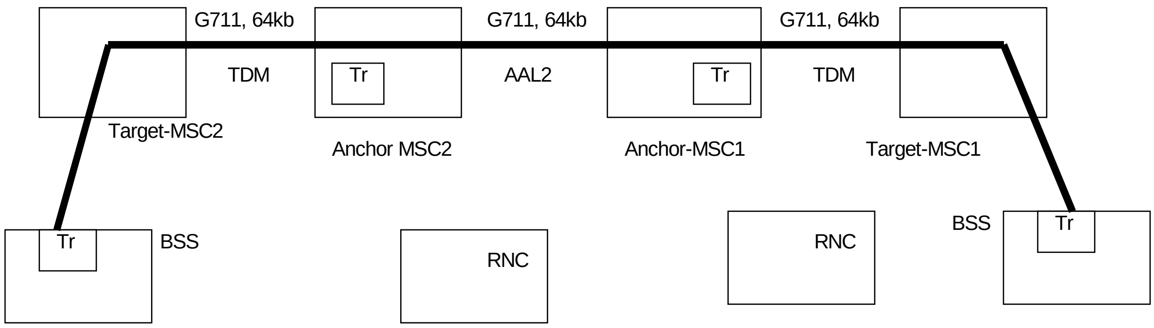


After inter-MSC handover at both sides toward 2G MSC with STM link we have the following configuration where 4 transcoders are present in the speech path end-to-end:



In such a situation the speech quality degradation may be unacceptable. Such a situation may also occur in case of handover to 3G MSC with TDM link.

For the transcoders to be inserted at the Target-MSCs side only and to remove them in the Anchor-MSCs a “Bearer Modification” is required between the two anchor-MSCs for having a 64kb AAL2 connection for G711. The configuration indicated in following figure is expected in future release:



But this requires BICC / QAAL2 CS2 for "Bearer Modification" capability handling. Besides, how both anchor MSCs know that they have to remove the transcoders is to be studied.

6.6 Information flow for sending a tone or an announcement

*[Editor's note: **This section needs further updates.** RAN3 plans to find suitable solution about this open issue in their March 2000 meeting. Thereafter, The description in this subclause will be reconsidered when RAN3 has decided on in-band or out-of-band solution. The possible candidate solutions are described for the time being.]*

[Alternative 1: Information flow for sending a tone]

A possible way out might be to perform the through connection in MSC-b only after MS-b has sent the Connect message (see fig.6.6/1). In this case, the RNCs start sending the Iu UP initialisation message as soon as the RAB assignment has been performed. However, the initialisation will not be answered by the opposite RNC until the connection has been established.

Note that in this case charging will be started before the Iu UP protocol has been established. This means, even if the Iu UP protocol cannot be established successfully, the subscriber will have to pay for the call. If this is acceptable to the operators, then at least care should be taken that the possible maximum time for which the RNC tries to establish the Iu UP protocol **after receipt of Connect** is limited effectively by some timer.

If the RAB assignment is performed before receipt of Alert, such an effective limitation is probably not possible without introducing a new message between MSC and RNC. Currently, the RNC does not know when the MS has sent Connect, as the DTAP signalling is transferred transparently. Therefore, the minimum time for which the RNC should retry the Iu UP protocol initialisation is given by the call control timer T301 which is started in the MSC with receipt of Alert and stopped with receipt of Connect. Its length is operator dependent; typical values are of the order of 1-3 minutes. This would also be the time for which the RNC would try to initialise the Iu UP protocol **after receipt of Connect**, if the subscriber B answered the call immediately after the alerting has started.

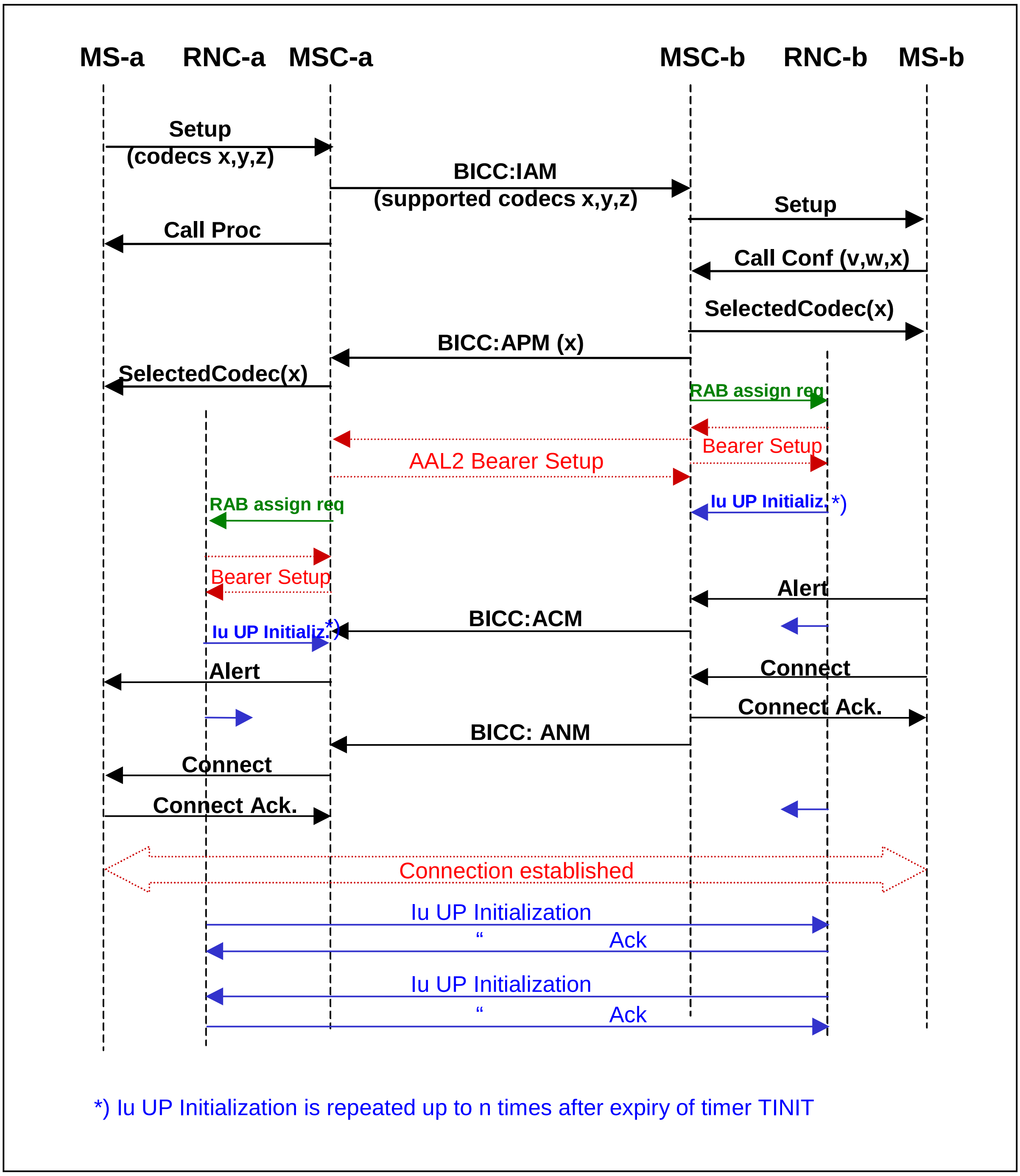


Figure 6.6/1: Information flow for applying RBT

[Alternative 2: Information flow for sending a tone]

Only DHO-DHO path is needed for the Iu UP initialisation. It is not necessary to complete the path to UE at destination side when Iu UP initialisation is performed. It is possible to inform RNC that the called user send back the answer message, "CONNECT", with new message so that RNC can connect RAB and Radio Bearer.

1. After the CC level negotiation, the bearer establishment for RAB and RB is performed. However RAB and RB at the destination side are not connected in RNC at this point.
2. When the called party answers, the destination MSC sends new message to RNC to indicate the answer from the called user.
3. On the receipt of the indication from MSC, RNC connects through the transmission path.
4. An answer message toward the originating MSC is sent, and then the charging may begin at the MSC controlling charging.

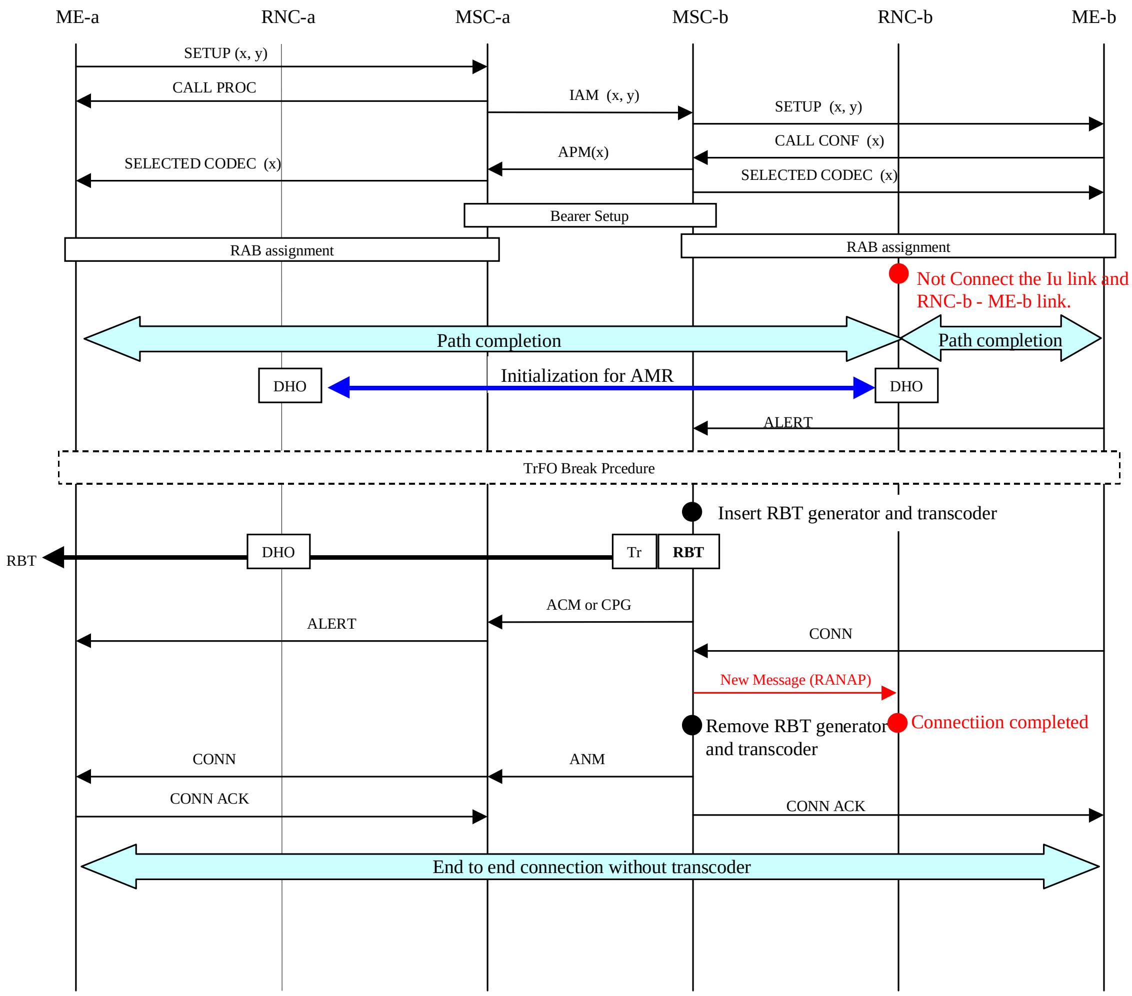


Figure 6.6/1: Information flow for applying RBT

7 Interactions with supplementary services

7.1 Call Deflection service (GSM 03.72)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclause 6.3 is applied.

7.2 Line identification services (GSM 03.81)

7.2.1 Calling Line Identification Presentation (CLIP)

No impact.

7.2.2 Calling Line Identification Restriction (CLIR)

No impact.

7.2.3 Connected Line Identification Presentation (COLP)

No impact.

7.2.4 Connected Line Identification Restriction (COLR)

No impact.

7.3 Call forwarding services (GSM 03.82)

7.3.1 Call Forwarding Unconditional (CFU)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclause 6.3 is applied.

7.3.2 Call Forwarding on mobile subscriber Busy (CFB)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclause 6.3 is applied.

7.3.3 Call Forwarding on No Reply (CFNRy)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclauses 6.3 is applied.

7.3.4 Call Forwarding on mobile subscriber Not Reachable (CFNRc)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclauses 6.3 is applied.

7.4 Call wait (GSM 03.83)

In order to apply the notice tone to the interjected party, the speech insertion procedure described in subclause 6.4 is applied.

7.5 Call hold (GSM 03.83)

In order to apply the notice tone to the held party, the speech insertion procedure described in subclause 6.4 is applied.

7.6 Multiparty (GSM 03.84)

In order to mix calls, the speech insertion procedure described in subclause 6.4 is applied.

7.7 Closed user group (GSM 03.85)

No impact.

7.8 Advice of charge (GSM 03.86)

No impact.

7.9 User-to-user signalling (GSM 03.87)

No impact.

7.10 Call barring (GSM 03.88)

7.10.1 Barring of outgoing calls

No impact.

7.10.2 Barring of incoming calls

No impact.

7.11 Explicit Call Transfer (GSM 03.91)

In case that a call A-B is transferred to C by B (A-C as result), A-B may use codec x, A-C may use codec y, the procedure described in subclause 6.3 is applied.

7.12 Completion of Calls to Busy Subscriber (GSM 03.93)

No impact.

8 Parameters

8.1 Codec type

The coding of the parameters belonging to a particular codec in UMTS is described in UMTS TS 26.103.

8.2 Codec list

The coding of the list of supported codecs in UMTS is described in UMTS TS 26.103.

9 Mapping of BC information on mobile radio interface layer 3 to codec type parameters for the BICC protocol and vice versa

The information elements and messages necessary to provide OoBTC are described in UMTS TS 24.008.

The parameters used on the mobile interface layer 3 protocol are described in UMTS TS 24.008. The parameters used on the BICC protocol are described in UMTS TS 26.103. Mapping of parameters given on the mobile radio interface layer 3 protocol to the BICC protocol (and vice versa) can be done one-to-one.

For the mapping of codec types to RAB QoS parameters, please see UMTS TS 25.413, UMTS TS 23.107, the UMTS TS describing that particular codec type, and UMTS TS 25.415.

10 Charging

The selected codec shall be included in all the call data records of the call legs involved in out-band codec negotiation belonging to a particular subscriber.

Annex A (Informative): Status of Technical Specification 23.153

Status of Technical Specification 23.153		
Date	**Version**	**Comments**
September 1999	0.0.0	First draft prepared by the rapporteur
October 1999	0.1.0	2nd draft prepared by the rapporteur (Updated version from Abiko.)
November 1999	0.2.0	3rd draft prepared by the rapporteur
December 1999	1.0.0	Submitted to CN#06 for information
February 2000	1.1.0	4th draft prepared by the rapporteur
February 2000	1.2.0	5th draft prepared by the rapporteur (Updated version from Milan.)
February 2000	1.3.0	6th draft prepared by the rapporteur (Updated version from Milan.)
February 2000	1.4.0	7th draft prepared by the rapporteur (Updated version from Milan.)
February 2000	1.5.0	8th draft prepared by the rapporteur (Updated version from Milan.)
March 2000	2.0.0	Submitted to TSG CN#07 for approval
Text and figures: **Stylesheet:** 3gpp_70.dot **Rapporteur**: Toshiyuki Tamura / Heinz-Peter Keutmann **Company:** NEC Corporation / Ericsson L. M.		

History

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