Presentation of Specification to TSG or WG

Presentation to: TSG CN Meeting #6

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Presented for: Information

Abstract of document:

The TS describes the technical realization of Out-of-Band Transcoder Control (OoBTC). OoBTC is a feature to enhance the speech quality and to save bandwidth in mobile-to-mobile calls. No transcoder shall be necessary for monitoring and controlling issues. The mobiles shall indicate supported codecs to the terminating MSC. The terminating MSC shall select a codec supported by both mobiles. The information about the selected codec is transported to the mobiles and the bearers are established accordingly to carry the call.

Changes since last presentation to TSG-CN WG2 Meeting #11:

None

Outstanding Issues:

The implementation is dependent on the finalization of the specification on coding of codec information (3G TS 26.103) in TSG-SA WG4 and on the Bearer-Independent Call Control (BICC) protocol in ITU-T SG11. Both specifications are expected to be ready by the end of the year.

Contentious Issues:

A new CC message might be needed to indicate the selected codec to the MS (TSG-CN WG1, CR to 3G TS 24.008).

Tdoc 3GPP N2-99H40

3GPP TSG-CN WG2

Phoenix, U.S.A.

15th – 19th November 1999

3G TS 23.153 V1.0.0 (1999-12)

Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Core Network;
Out of Band Transcoder Control - Stage 2;
(3G TS 23.153 version 1.0.0)

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Reference

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Contents

Intellectual Property Rights

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

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

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, data, or multimedia in order to utilize 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] UMTS TS 23.107: "QoS Concept and Architecture"

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

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

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

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

[6] ITU-T AB.CDE: “Speech codec list for GSM and UMTS”. *Editor’s Note: Numberand title not assigned yet.*

[7] Q.BICC: "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.

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: A transcoder device is physically present in the signal path, but the transcoding functions are bypassed. The transcoding device may perform control and protocol conversion functions.

Transcoder: A transcoder is a device to change the encoding of information from one particular encoding scheme to a different one.

Transcoder Free Operation: No transcoder device is physically present and hence no control or conversion or other functions can be associated with it.

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 Functionality

4.1 Required functionality

- The capability to negotiate the preferred codec type to be used between two end nodes and to avoid two transcoders in the network.
  - The originating UE may indicate the list of supported codecs for codec negotiation. This list shall be conveyed to the terminating MSC through the networks to determine the codec type to be used and establish the transcoder-less connection.
  - The terminating UE may indicate the list of supported codecs to the terminating MSC.
- The capability to control (avoiding and reverting) transcoders in the network.
  - Avoiding transcoders:

    Intervened transcoders in the network should be avoided at any time depending on the result of transcoder negotiation procedure and the situation of the Call State. Mostly, the followings are typical cases for avoiding transcoders:

    - The transcoder-less connection was reverted to the normal call connection by some reasons, then returns back to the Call State that can configure the transcoder-less connection again. (Ex. Multiparty call returns to the simple A to B call connection.)
  - Reverting to the normal call connection from the transcoder-less connection:

    If the call connection encounters the following situations, the transcoder-less connection is reverted to the normal call connection.

    - SS interruptions (Ex. to B call connection becomes to multiparty call connection.)
    - DTMF signal is detected.
- The codec types comprise codecs for speech, data, and multimedia. The transcoder control should have enough expandability to support future enhancements of codec types.
- The transcoder control procedure should be independent from the location of the transcoder in the network.
- The transcoder control procedure should not cause a perceivable time lag in the cases of establishing transcoder-less connection and reverting to normal call connection.
- The transcoder-less connection should be maintained if the UE executes hand-over.

Note: For a codec supporting various modes, the described functionality shall also be applicable to negotiate the set of codec modes common to originating and terminating UE.

4.2 Relationship between OoBTC and In-band TFO

Basically both OoBTC and In-band TFO are used for to establish the UE-UE though connection. However, each procedure is independent and does not interwork with the other one. OoBTC intentionally tries to make the TrFO so that the process for OoBTC performs during the call-establishment phase. Instead, the in-band TFO is activated only if transcoders located in each end node enable communication to each other so that the process for in-band TFO performs after the call connection has been established.

If the OoBTC fails to establish the TrFO and locates transcoders to the two end nodes, in-band TFO may perform afterward. Therefore, the In-band TFO is an appropriate technique for interworking with the 2G system (i.e. GSM) unless OoBTC is deployed in the 2G system.

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. (One decoder is to monitor one UE.)

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. Most calls traverse multiple CC nodes in one or more networks, and for speech calls some transit CC nodes may need to have access to the user plane (UP) information in order to perform a series of functions such as introduction of tones or announcements, voice prompting, etc. Therefore, these transit CC nodes need to understand the format of the UP speech in order to be able to transcode whenever needed. 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, analyze 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: analyze 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.

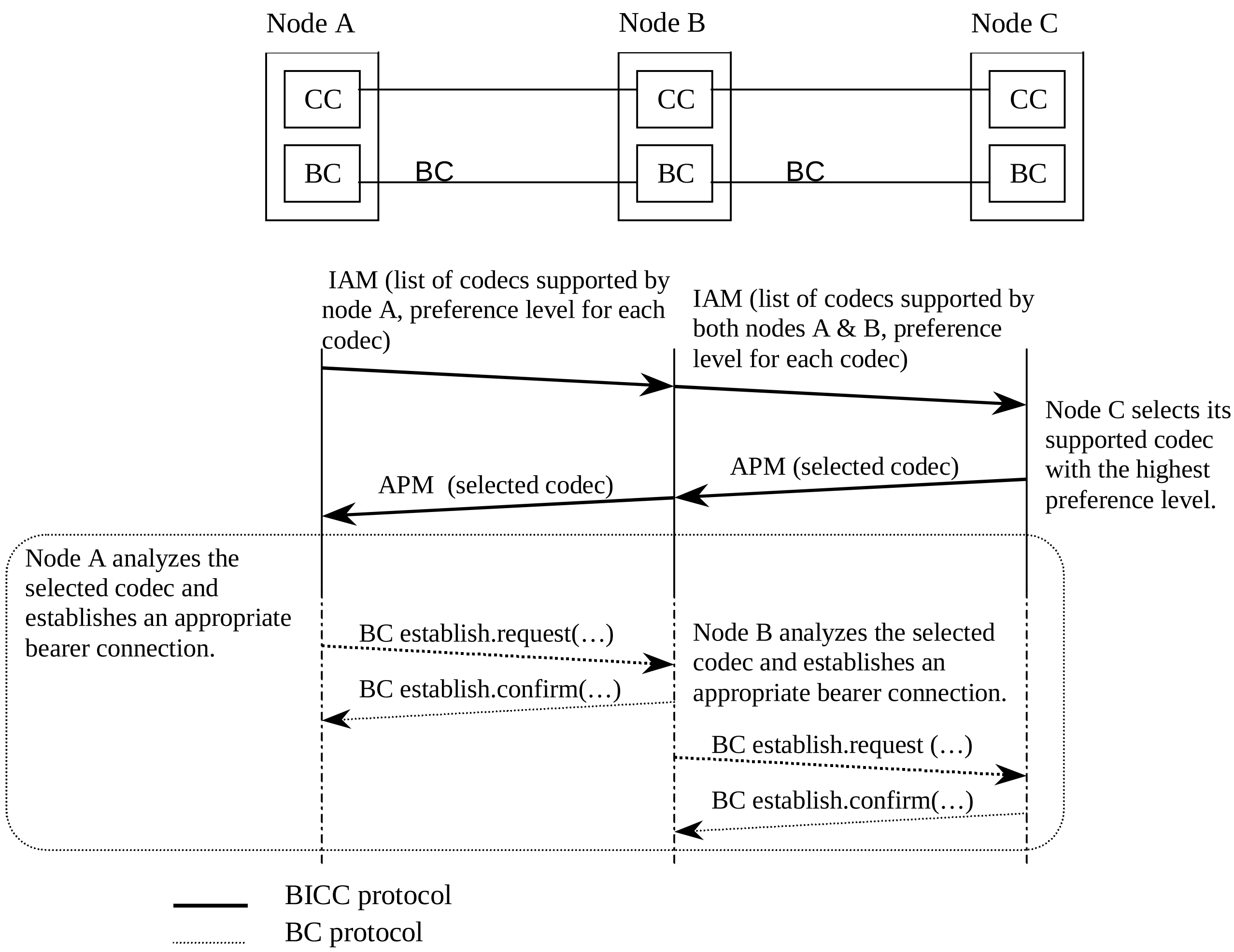

Figure 5.1/1. Sequence of BICC and BC messages for the proposed codec negotiation

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.

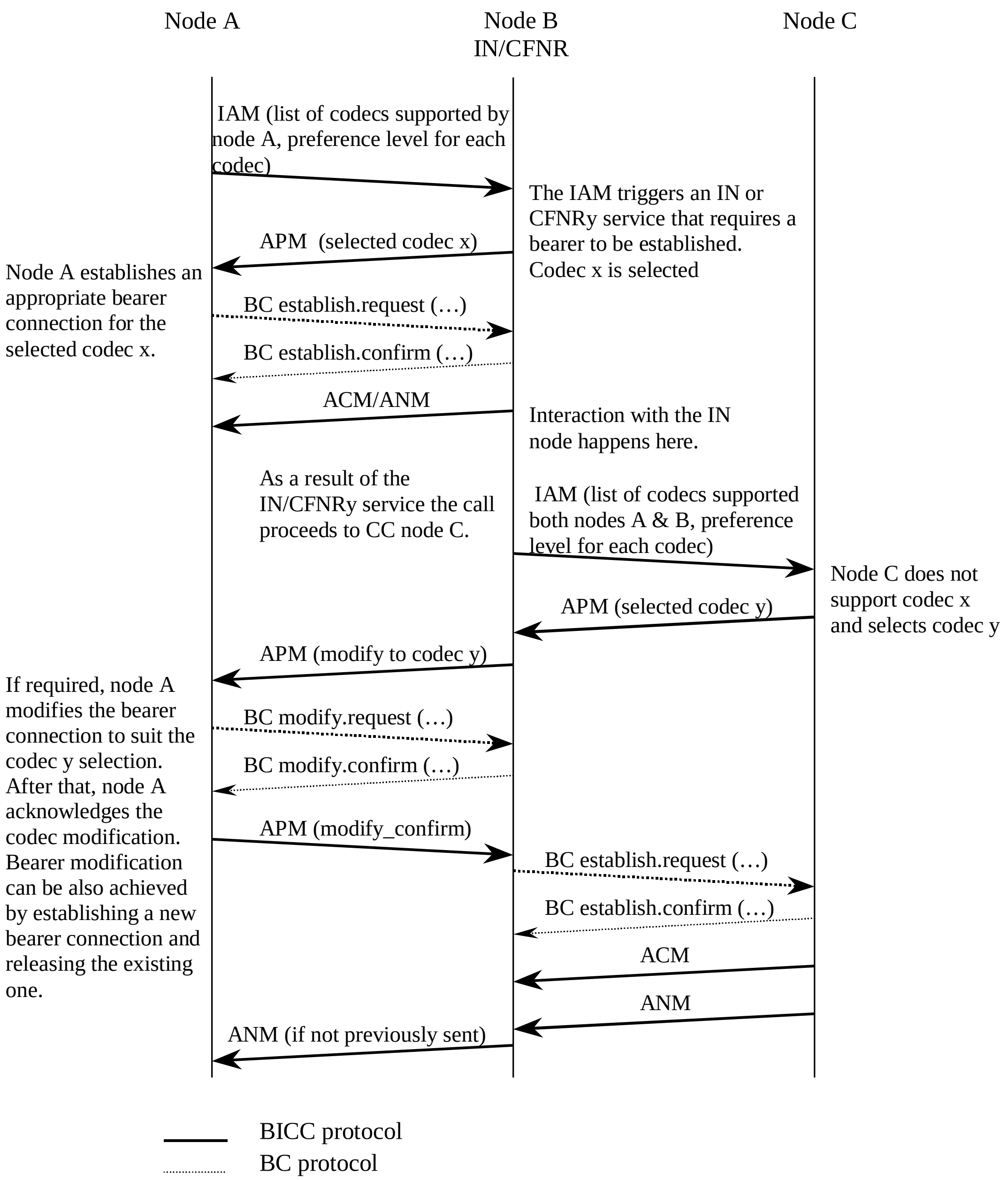

Figure 5.3/1. Sequence of BICC and BC messages for case of IN prompting services and CFNRy

5.4 Conference calls

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.

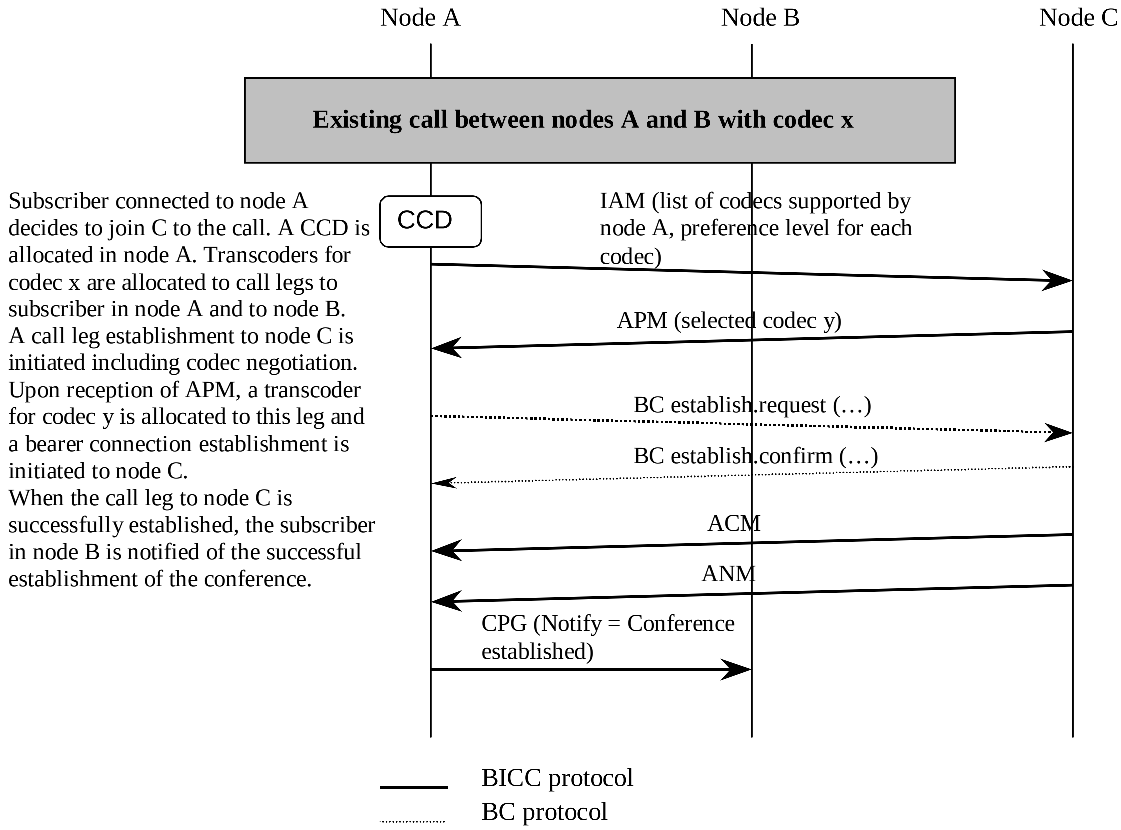

Figure 5.4/1. Sequence of BICC and BC messages for the case of conference calls.

5.5 Interworking with ISDN/PSTN

A third case for the codec negotiation arises when the BICC protocol interworks with the ISUP protocol. In this scenario, the interworking node has to temporarily terminate the codec negotiation in order to guarantee that a UP connection is established before forwarding the IAM inside the ISUP network. The interworking node shall select a codec from the received list of codecs and allocate a TRAU or rate adaptation unit (bit rate of compressed speech to 64 kbps used in ISDN/PSTN) so that the call establishment can proceed inside the ISDN/PSTN.

Figure 5.5/1 illustrates the signalling flow when the ISUP in node C either supports the APM and the APM-user or it is able to convey this information to node D by means of ISUP's compatibility mechanism. If this is not the case, the call will proceed inside the ISDN/PSTN without any out-of-band negotiation at all. In this example, ISUP in node D supports the APM-user.

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

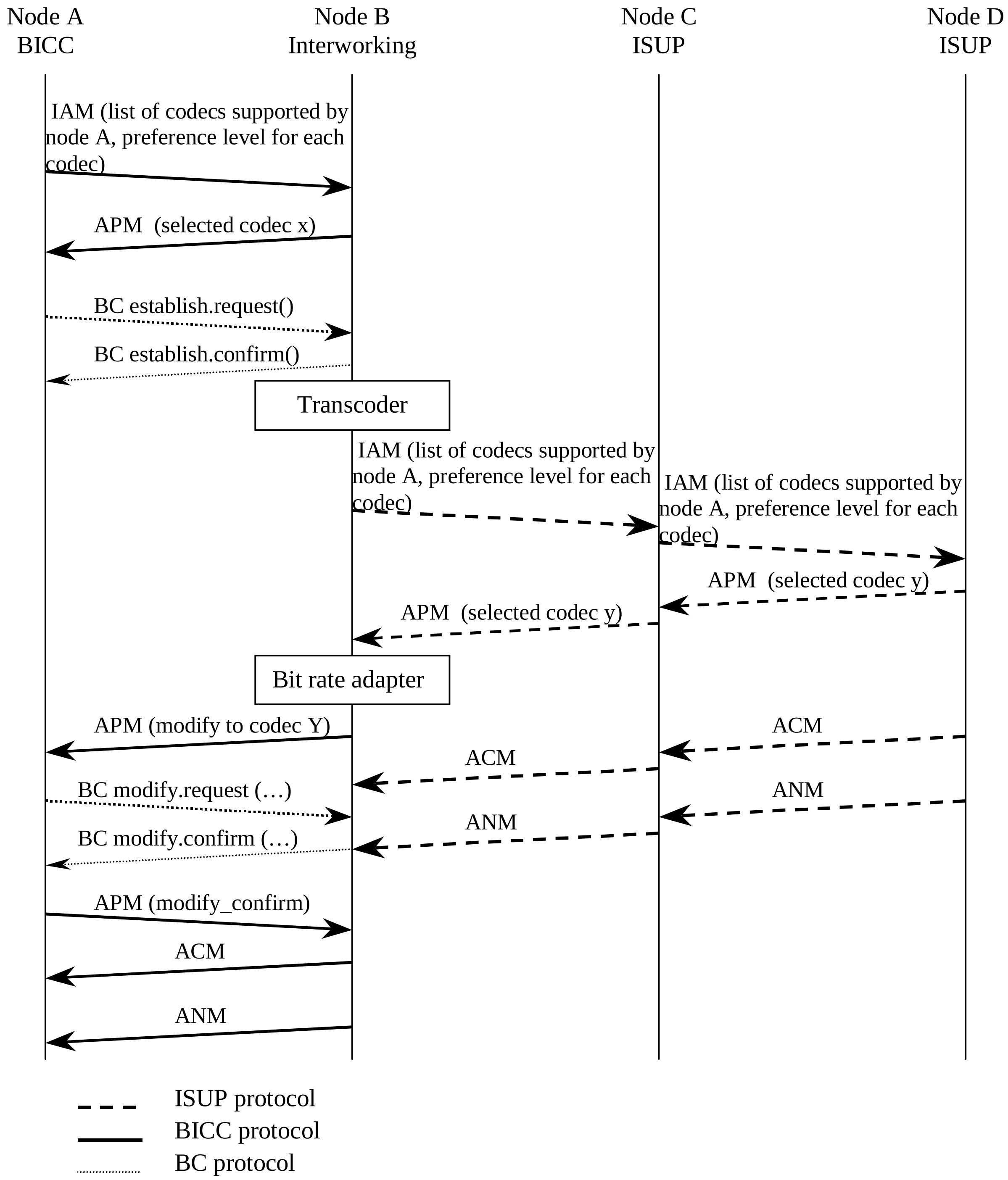

Figure 5.5/1. Sequence of BICC, ISUP and BC messages for the BICC/ISUP interworking scenario

6 Information flows

6.1 Information flows for MO calls

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

6.1.1 Successful MO call

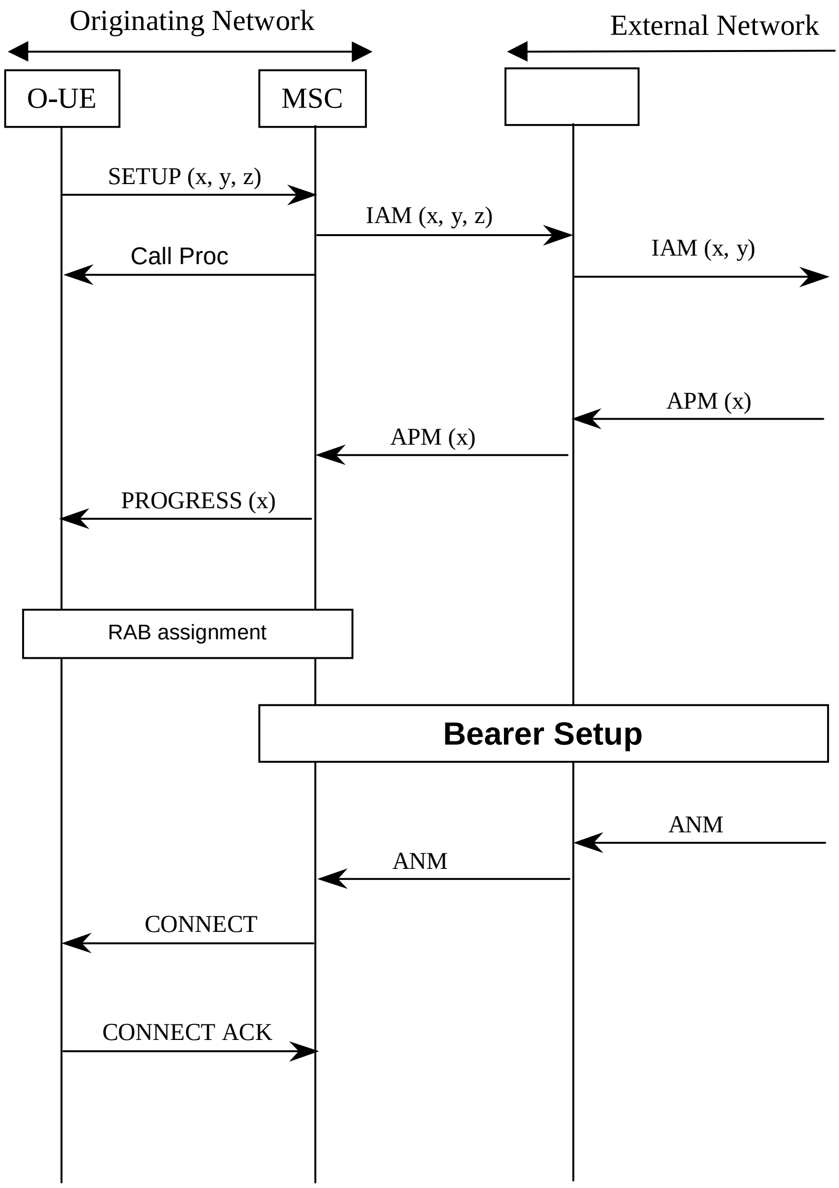

Figure 6.1.1/1 Successful MO call

- 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 was selected as the bearer to be used 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. The bearer has to be set up after receiving the APM (codec x) message because the size of the bearer is dependant upon the codec selected.

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

6.1.2 Unsuccessful MO call due to transit network not supporting BICC

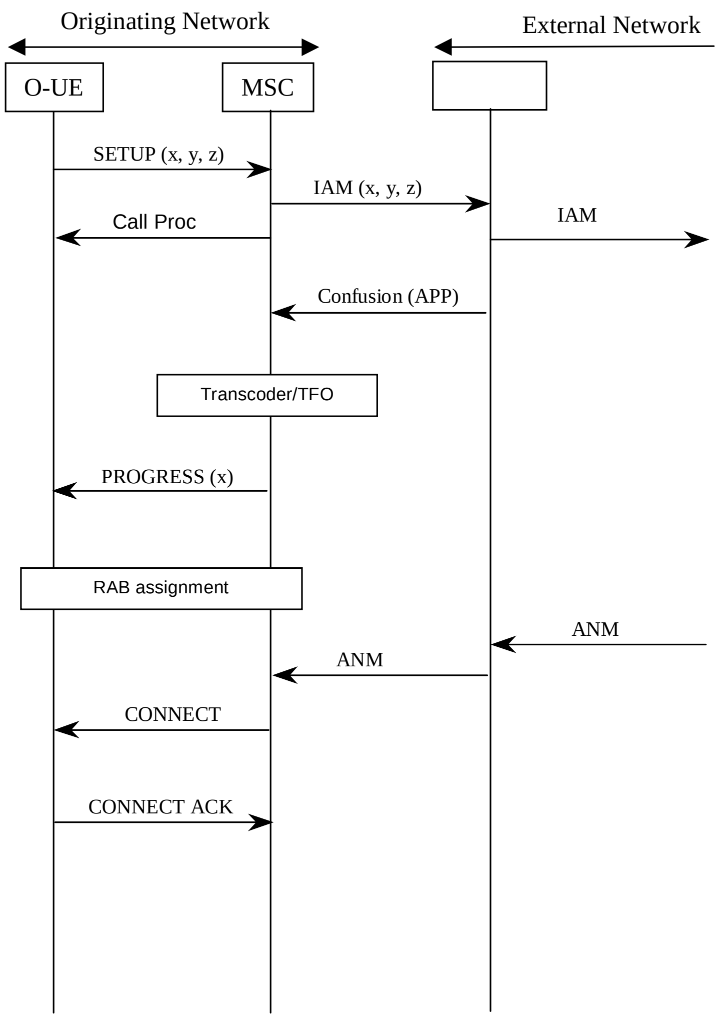

Figure 6.1.2/1 Unsuccessful MO call due to transit network not supporting BICC

- 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. Refer to the subclause 9 for mapping rules.
- If unrecognised information is not allowed in the external network, the entry node will follow the “pass-on not possible” indicator and send a “confusion” message to the MSC in originating network indicating that the APP parameter was discarded. Upon reception of the “confusion” message 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 PROGRESS message, the O-UE is informed that codec x has been selected, and RAB assignment will occur.

6.1.3 Unsuccessful MO call due to APM-user not supported

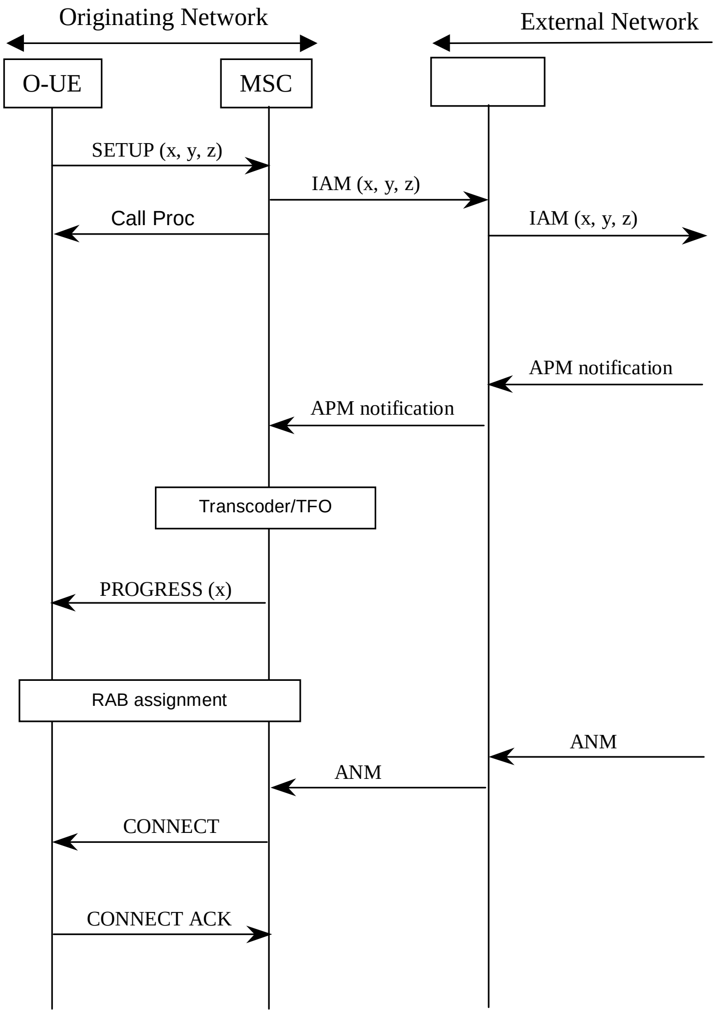

Figure 6.1.3/1 Unsuccessful MO call due to APM-user not supported

- 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. Refer to the subclause 9 for mapping rules.
- On reception of the APM notification due to some reasons. Ex.) APM-user is not supported in the terminating node. In this case, the MSC knows that out-of-band codec negotiation is not possible end-to-end and a transcoder is allocated.
- In the PROGRESS message, the O-UE is informed that codec xhas 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 MT call establishment scenarios.

6.2.1 Successful MT call

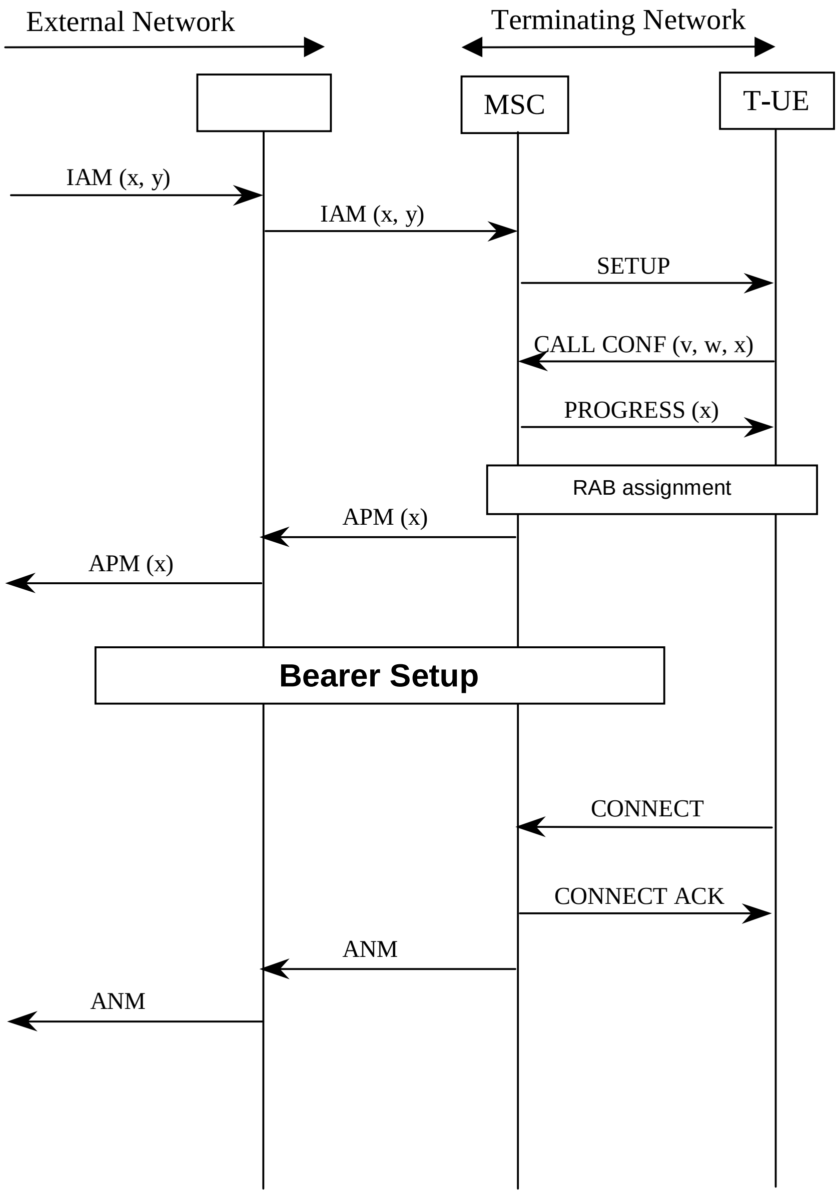

Figure 6.2.1/1 Successful MT call

- 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 PROGRESS 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 Unsuccessful MT call due to APM-user not supported

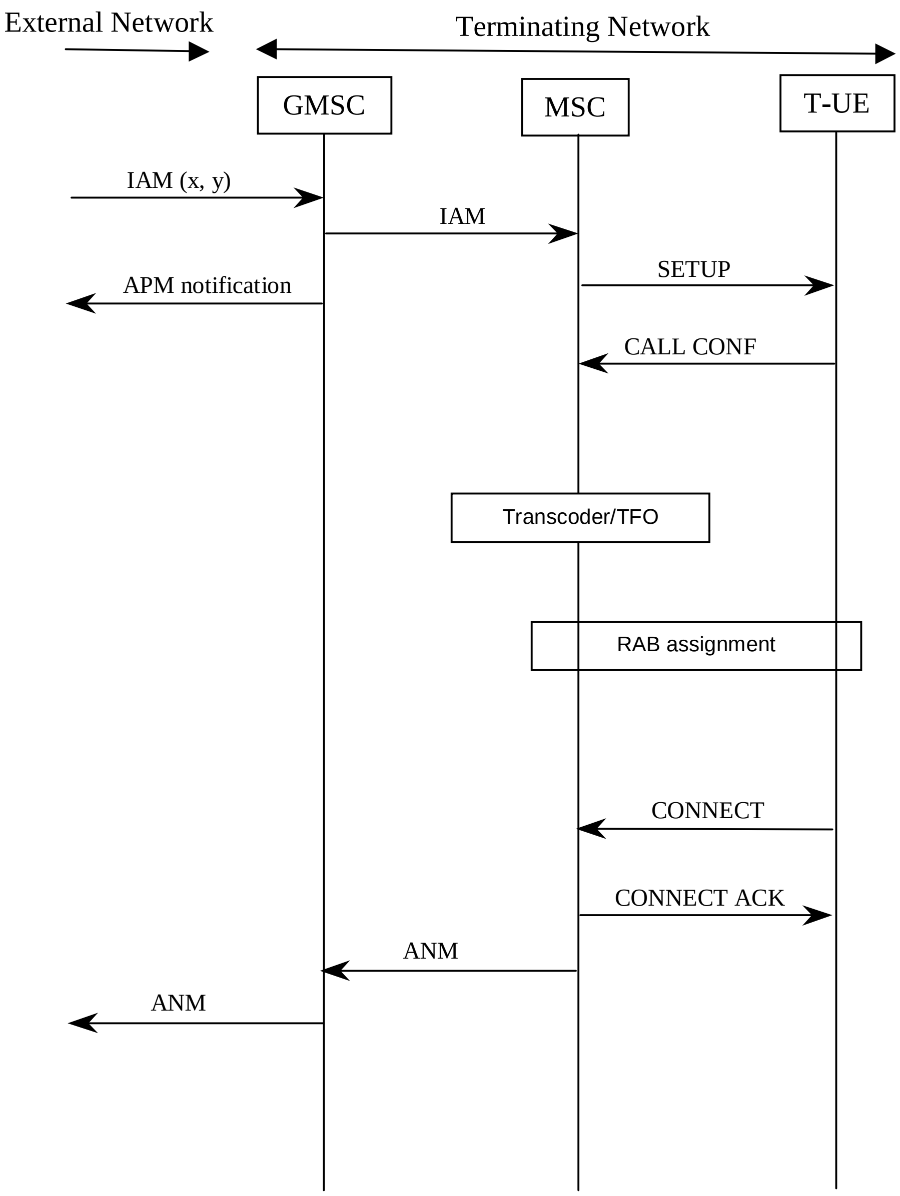

Figure 6.2.2/1 Unsuccessful MT call due to APM-user not supported

- 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 GMSC 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

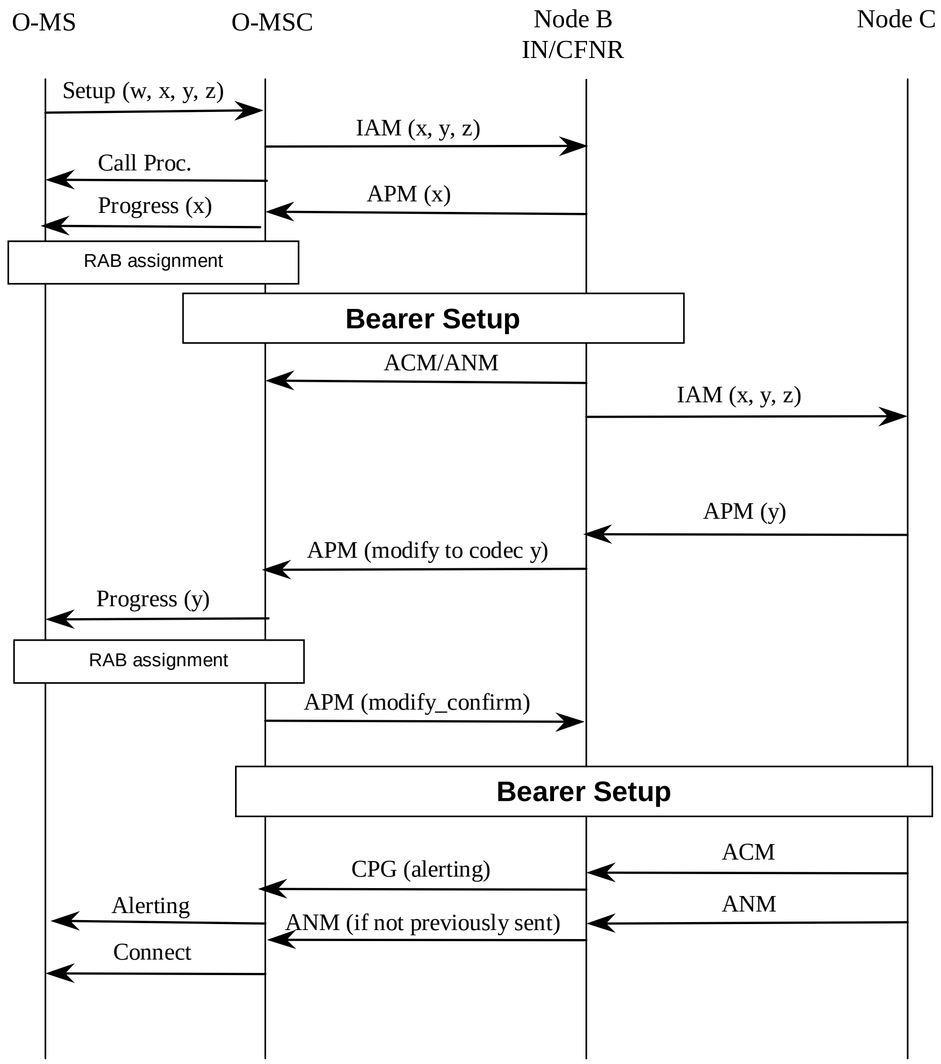

Figure 6.3/1 Interactions with IN or CFNR SS at call setup

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 5.3/1 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.

6.4 Information flow for interaction with Multiparty SS

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.

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.

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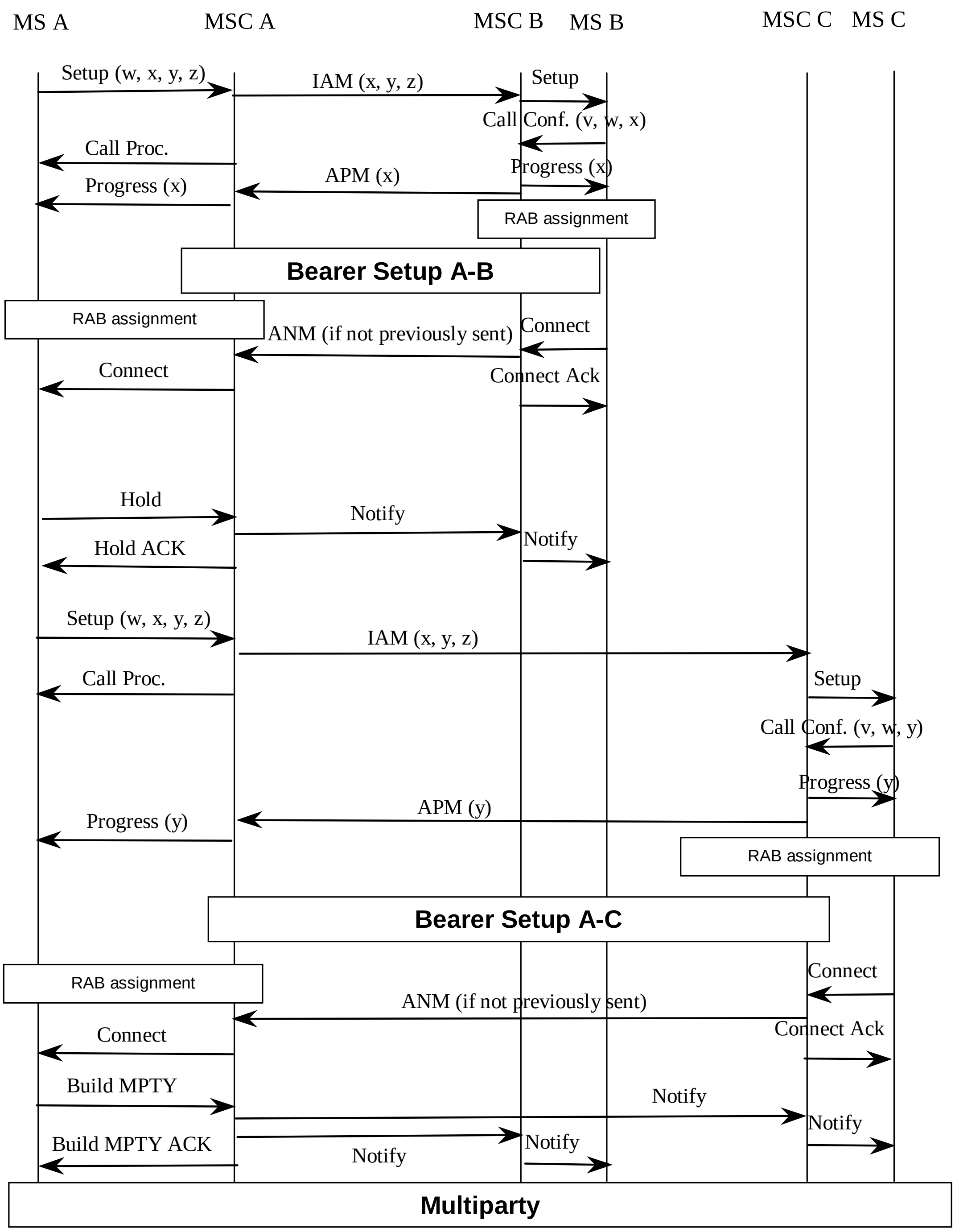

Figure 6.4/1 Interactions with Multiparty SS

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

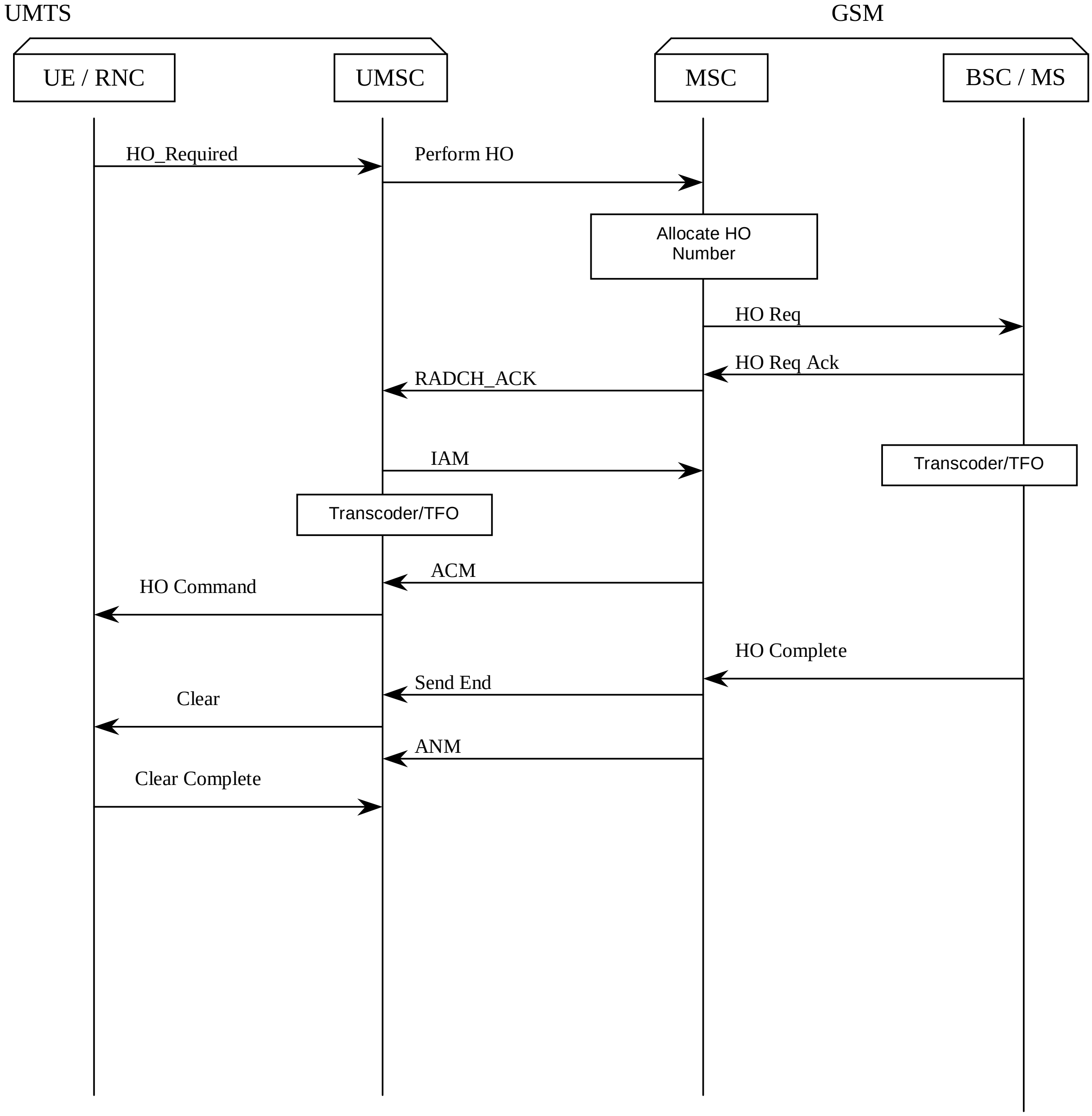

Figure 6.5/1 UMTS to GSM Handover after TrFO establishment

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 will transcode to PCM. The GSM BSC will perform transcoding in the same manner, which currently used in GSM. TFO can be used to reduce quality degradation.

If the transport link between the UMSC and MSC is not TDM (e.g. AAL2 is supported), then the codec negotiation between the UMSC and MSC will be performed using the procedures shown in subclause 6.1.1 and subclause 6.2.1. In this case, the MSC will transcode from low bit rate speech to PCM across the GSM A-interface. The BSC will order the transcoding according to normal operation.

7 Interactions with supplementary services

7.1 Call Deflection service (GSM 03.72)

No impact.

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 isertion 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)

No impact.

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 (see UMTS TS 24.008) are described in UMTS TS ab.cde. The parameters used on the BICC protocol are described in ITU-T AB.CDE (which is the ITU-T mirror of 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	0.2.0	Submitted to CN#06 for information
Text and figures: **Stylesheet:** 3gpp_70.dot **Rapporteur**: Toshiyuki Tamura / Heinz-Peter Keutmann **Company:** NEC Corporation / Ericsson L. M.		

History

Document history		