3GPP TSG_CN#6 **NP-99xxx**
Nice, France
13th – 15th December 1999

Source: MCC
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Title: N2B Output Documents from Phoenix (Nov. 99) – Agreed Liaison Statements

Agenda item:

Document for: Information

N2B agreed to the following LS

	Work Item	Tdoc N2-99	Destination	Titles
1.	OoBTC	J56	To: S4 CC: S2	LS to S4 cc:S2 on TFO within GSM
2.	OoBTC	J58	To: S2, S4	LS to S2, S4 on status of OoBTC
3.	OoBTC	J60	To: ITU-T SG11	LS towards ITU-T SG11 on codec information
4.	Security	J86	To: S3	LS to S3 on Confirmation of TS 33.102 section 6.3.4 Distribution of authentication vectors between VLRs
5.	GPRS	J78	To: S2	LS to S2 on DHCP end-to-end and Mobile IP support
6.	Handover	J84	To: S1, S2	LS to S1, S2 on Open Issues on 3G to 2G PS domain Handover
7.	Follow Me	J90	To S2	introduction of Follow Me - interfaces
8.	QoS	K04	To S2	LS to S2 on QoS Task force
9.	Combined MM	K66	To S2 Cc S1	LS to S2 on Combined MM

3GPP TSG CN2 Meeting #12 **Tdoc N2-99J56**
Phoenix, AZ, USA
15 - 19 November, 1999

Subject: **LS on Out-of-Band Transcoder Control**

To: **TSG-S4**

CC: **TSG-S2**

From: **TSG N2**

N2 thanks S4 for reviewing the technical report on Out-of-Band Transcoder Control (OoBTC).

N2 has received S4's LS (TSGS4#6(99) 267) identifying the issues S4 has found.

After reviewing the LS, N2 realized that S4 might interpret OoBTC to be applicable to both GSM and UMTS. However, this interpretation would not be correct.

According to N2, OoBTC targets UMTS in R99 only.

The scope of OoBTC might be expanded in future to cover GSM as well. This could be done earliest for R2000, provided that a WI for the expansion of OoBTC to GSM is agreed.

3GPP TSG CN2 Meeting #12 **Tdoc N2-99J58**
Phoenix, AZ, USA
15 - 19 November, 1999

Subject: **LS to S4 on Out-of-Band Transcoder Control**

To: **TSG-S4**

CC: **TSG-S2**

From: **TSG N2**

N2 thanks S4 for reviewing the TR on Out-of-band transcoder control (OoBTC) and providing the LS (TSGS4#6(99) 267) to identify the issues that S4 has found.

The following text gives brief explanations to the issues raised. The TS describing the technical realization of OoBTC has been updated to cover all concerns.

N2 should be grateful if S4 would review the attached revision of the TS on OoBTC stage 2 to ensure that all open questions have been settled.

N2 would like to inform S4 that the stage 2 description of OoBTC has been raised to version 1.0.0 and shall be presented to N plenary for information.

To continuously progress the work on OoBTC, N2 would appreciate receiving feedback to their next meeting 17-21 January 2000.

Happy holidays!

- The report focuses on the signaling required to establish a Transcoder Free configuration. S4 believe that an out-of-band signaling solution should also offer the possibility to establish a Tandem Free Operation where a transcoder is inserted in the communication path but the transcoding functions are bypassed. This solution could be beneficial when TFO must be discontinued during the call (DTMF, Multiparty call activated…). In that case, the transcoder would only have to monitor the synchronization and revert to normal transcoding if commanded by the Core Network or when TFO is discontinued.

Answer: As mentioned in the feasibility study of OoBTC (which is reflected in the TR on OoBTC), N2 has concluded that a major drawback of TFO consists in the necessity to allocate a transcoder for every call. Allocation of a transcoder that will be used only temporarily has been rated as a disadvantage. In other words, the transcoder as the hardware resource shall be inserted into a call only if needed. Therefore, N2 does not agree with the S4 concerns at least for R99.

- The 3G TR ab.cde does not address the synchronization monitoring. How will that be performed in Transcoder Free Operation?

Answer: Since no monitoring instance shall be present in a call using OoBTC, N2 expects that the involved codecs shall need to perform synchronization on their own.

- In case of Transcoder Free Operation, which format will be used to convey the speech data? What is the compatibility with GSM?

Answer: In GSM, PCM was used to transport the speech data. Using the BICC protocol in

UMTS for OoBTC, the transport format is determined by the bearer to which the BICC protocol is applicable to (e.g. AAL2, IP). In the case interworking with GSM requires a change of the bearer, transcoding is necessary.

- How could Transcoder Free Operation be implemented on an existing GSM network, providing that the TRAU is part of the BSS? or should the Core Network be equipped with internal Transcoders in that case?

Answer: N2 targets to standardize OoBTC technique only for 3G at least in R99; therefore the OoBTC working in 3G does not interwork with GSM in R99. In addition, section 4.2 in TS describes the relationship between OoBTC and In-band TFO. For an expansion of OoBTC to GSM as well, a WI to study the impacts would be needed. At a very first glance, a possible solution could look "similar" to TFO: the compressed speech information would be passed unchanged through the BSS to be transported via the PCM-link to the terminating BSS to be passed on to the MS. In-depth study would be needed to assess the impacts of e.g. Multiparty SS, handover, insertion of tones or announcements.

- How will the management of different AMR configurations (i.e. Active Codec Modes, Supported Codec Modes, Maximum number of codec modes) be done? What is the impact on the BICC specification?

Answer: To deal with different AMR configurations, the procedure to negotiate supported codecs shall be applied accordingly: The originating UE shall indicate its supported codec modes (of a particular codec) in addition to the supported codec types. This information shall be transported from the originating to the terminating MSC. The terminating UE shall indicate – in the same way – its supported codec modes in addition to the supported codec types. From both lists, the terminating MSC determines the selected codec. If the selected codec comprises various codec modes, the intersection of the set of supported codec modes from both the originating UE and the terminating UE gives the set of active modes. When signalling the selected codec to both the terminating and the originating UE, this set is included in the information describing the selected codec. Since for UMTS the maximum number of supported codec modes is not limited (i.e. the UMTS AMR codec comprises 8 different codec modes and the maximum number of active codec modes equals to 8 as well), no further action is needed. If the maximum number of active codec modes should be subject to limitation and the number of supported codec modes in the intersection exceeds that limit, only those codec modes offering the best speech quality shall be selected from the intersection. No impact on the BICC specification is foreseen since the intended coding of codec parameters in BICC caters for that situation as well.

- How will the in-band signaling used in GSM for the AMR link adaptation (see GSM 05.09) be treated in case of Transcoder Free Operation between GSM and 3G?

Answer: Since N2 targets to standardize OoBTC technique only for 3G at least R99, therefore the OoBTC working in 3G does not interwork whith GSM in R99. In addition, section 4.2 in TS describes the relationship between OoBTC and In-band TFO.

- As for In-Band TFO, downlink DTMF will break Transcoder Free Operation and will be subject to the non-ideal transparency performances of the speech codec. S4 believe that out-of-band Layer 3 signaling for downlink DTMF should be considered.

Answer: N2 recognized that the downlink DTMF is related to the outband L3 signaling not related to the OoBTC. Therefore N2 concluded that this concern does not have any impact to the OoBTC. In addition, the DTMF signal detection is treated as the trigger to revert to the normal call if transcoder-less connection has been established. It is described in section 4.1 in TS as the required functionality.

Tdoc 3GPP N2-99H40

3GPP TSG-CN WG2

Phoenix, U.S.A.

15th – 19th November 1999

3G TS ab.cde V0.2.0 (1999-11)

Technical Specification

3rd Generation Partnership Project; Technical Specification Group Core Network; Out of Band Transcoder Control - Stage 2; (3G TS ab.cde version 0.2.0)

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Reference

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Contents

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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

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.

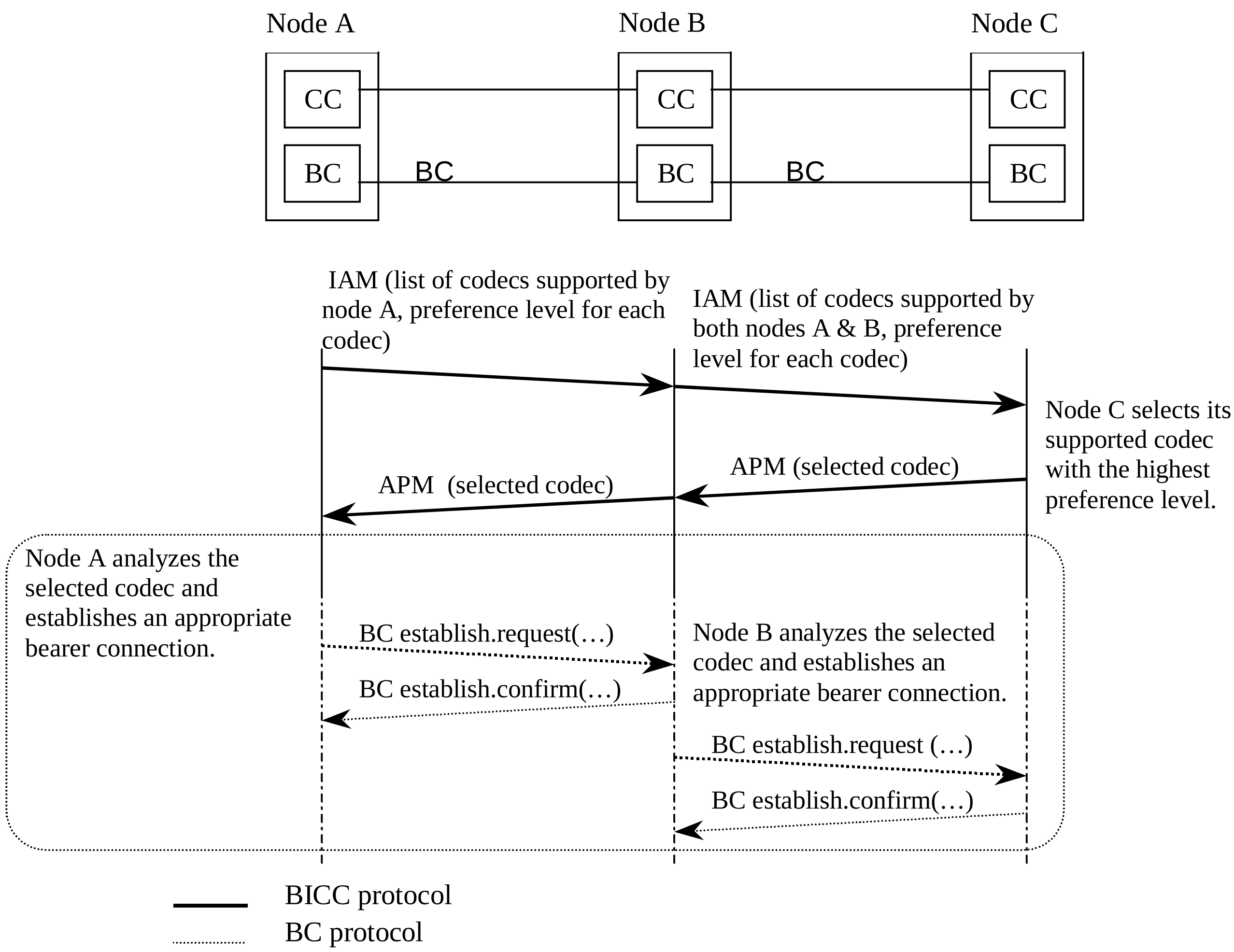

Figure 5.1/1. Sequence of BICC and BC messages for the proposed codec negotiation

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.

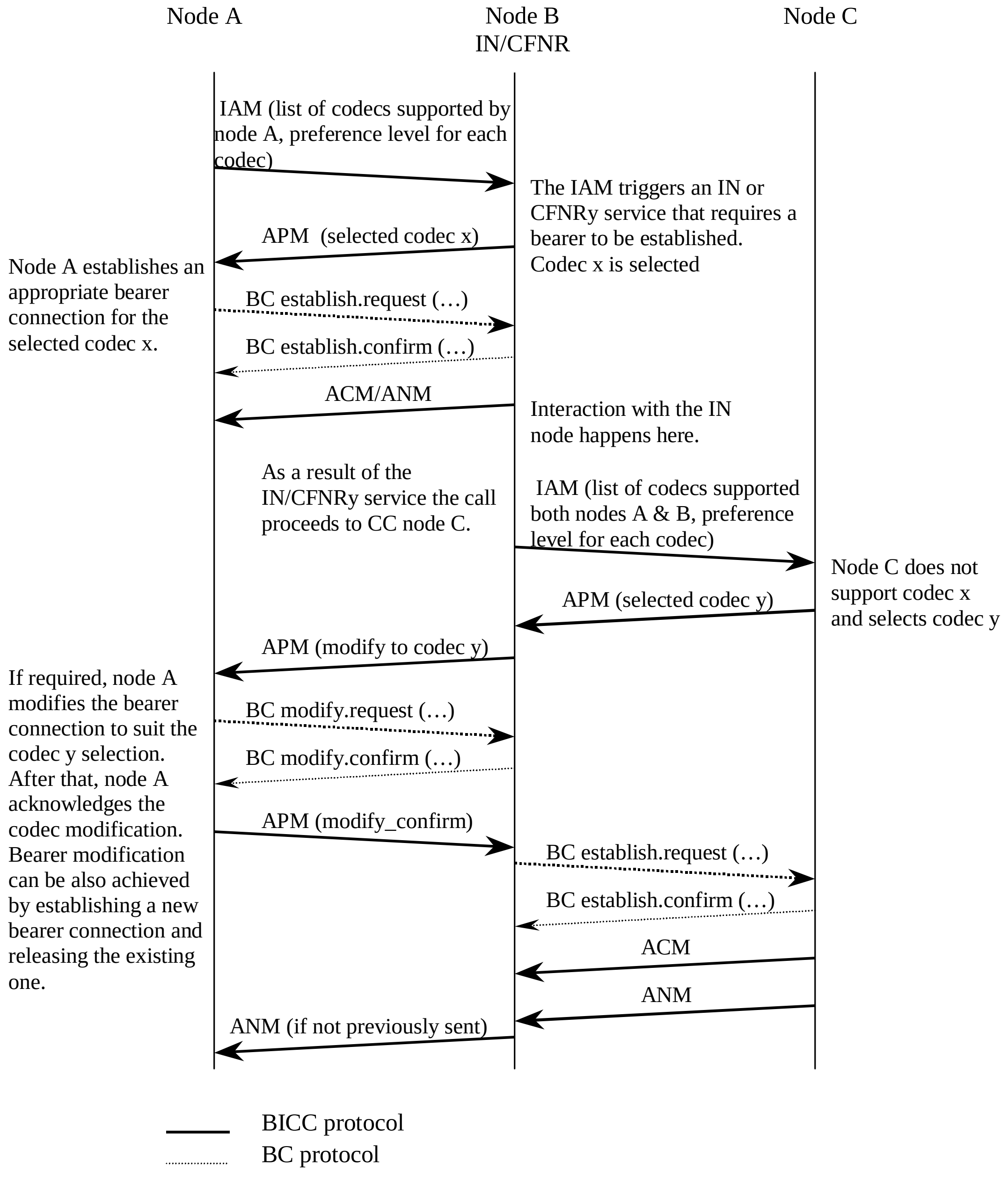

Figure 5.3/1. Sequence of BICC and BC messages for case of IN prompting services and CFNRy

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.

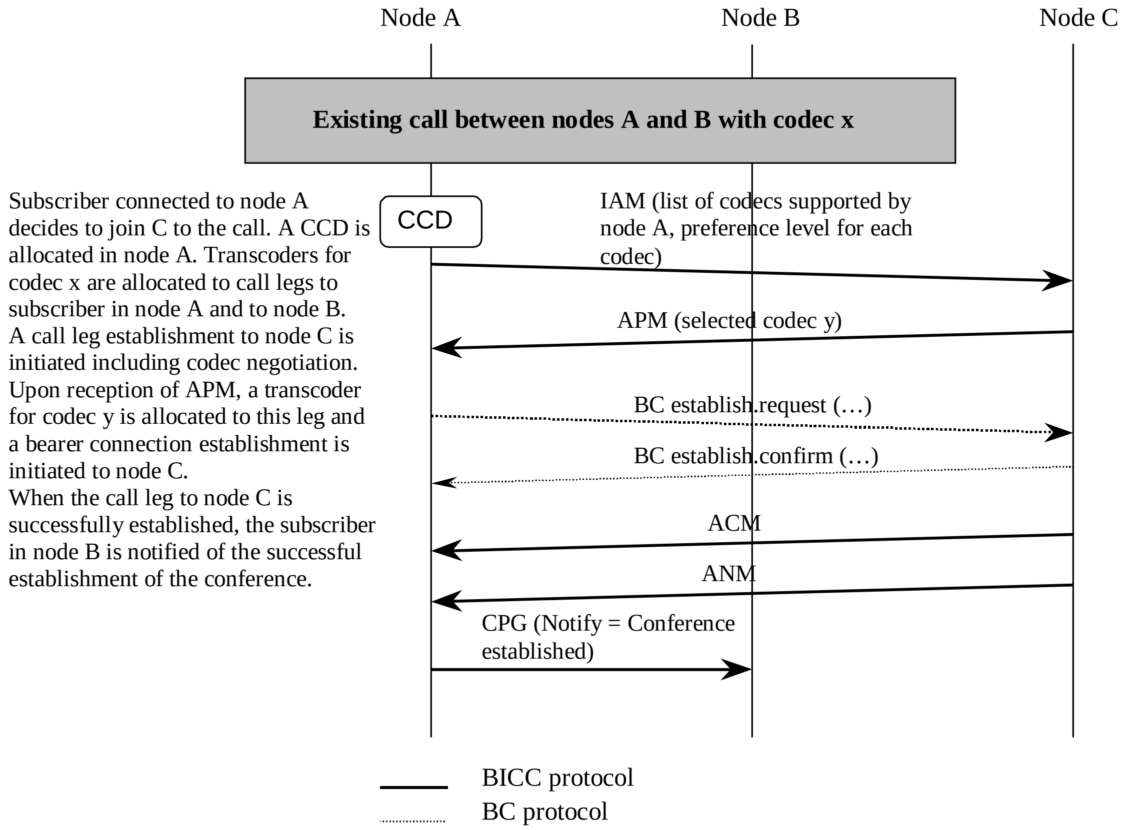

Figure 5.4/1. Sequence of BICC and BC messages for the case of conference calls.

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.

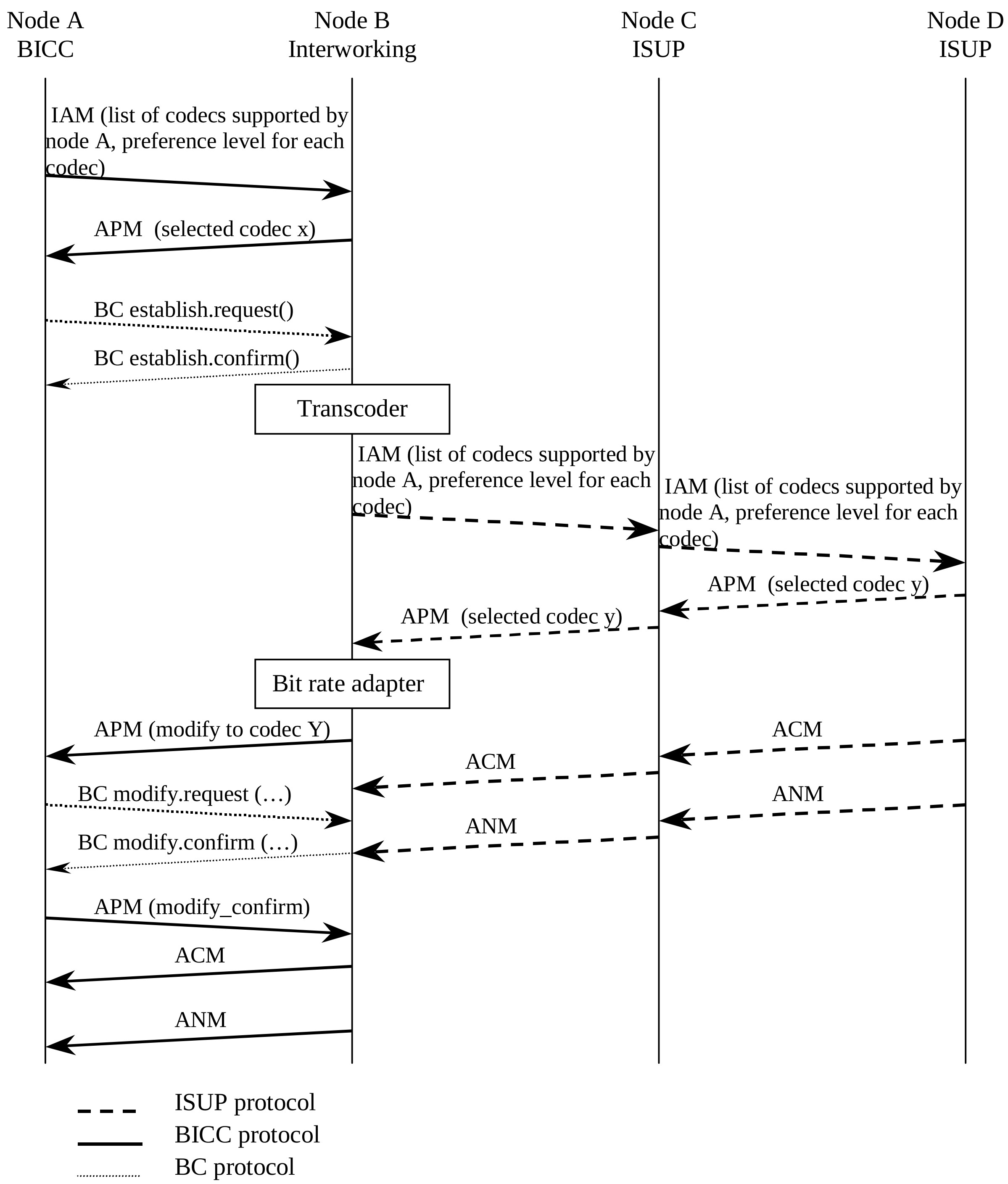

Figure 5.5/1. Sequence of BICC, ISUP and BC messages for the BICC/ISUP interworking scenario

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

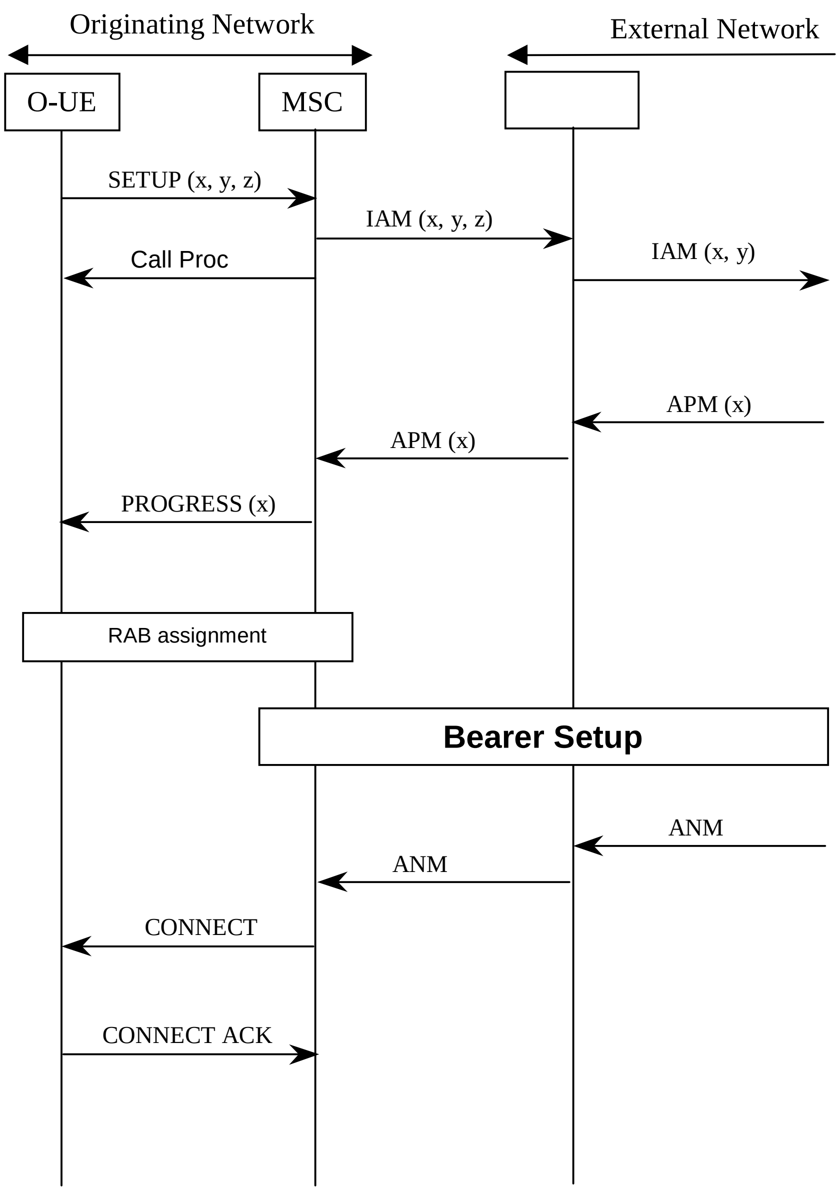

Figure 6.1.1/1 Successful MO call

- 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

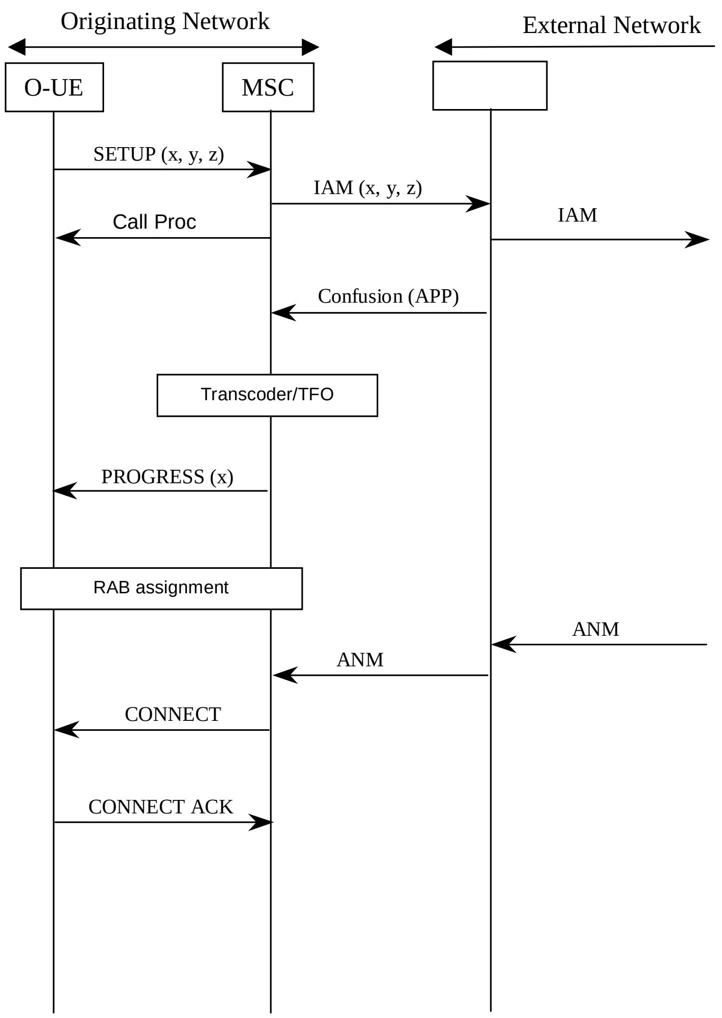

Figure 6.1.2/1 Unsuccessful MO call due to transit network not supporting BICC

- 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

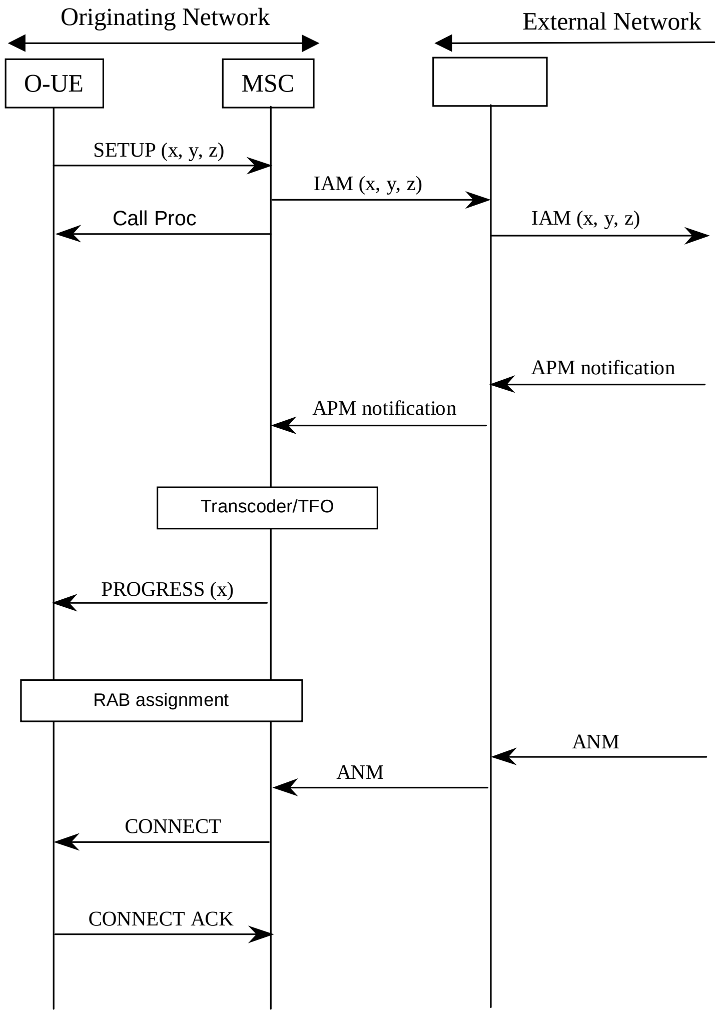

Figure 6.1.3/1 Unsuccessful MO call due to APM-user not supported

- 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

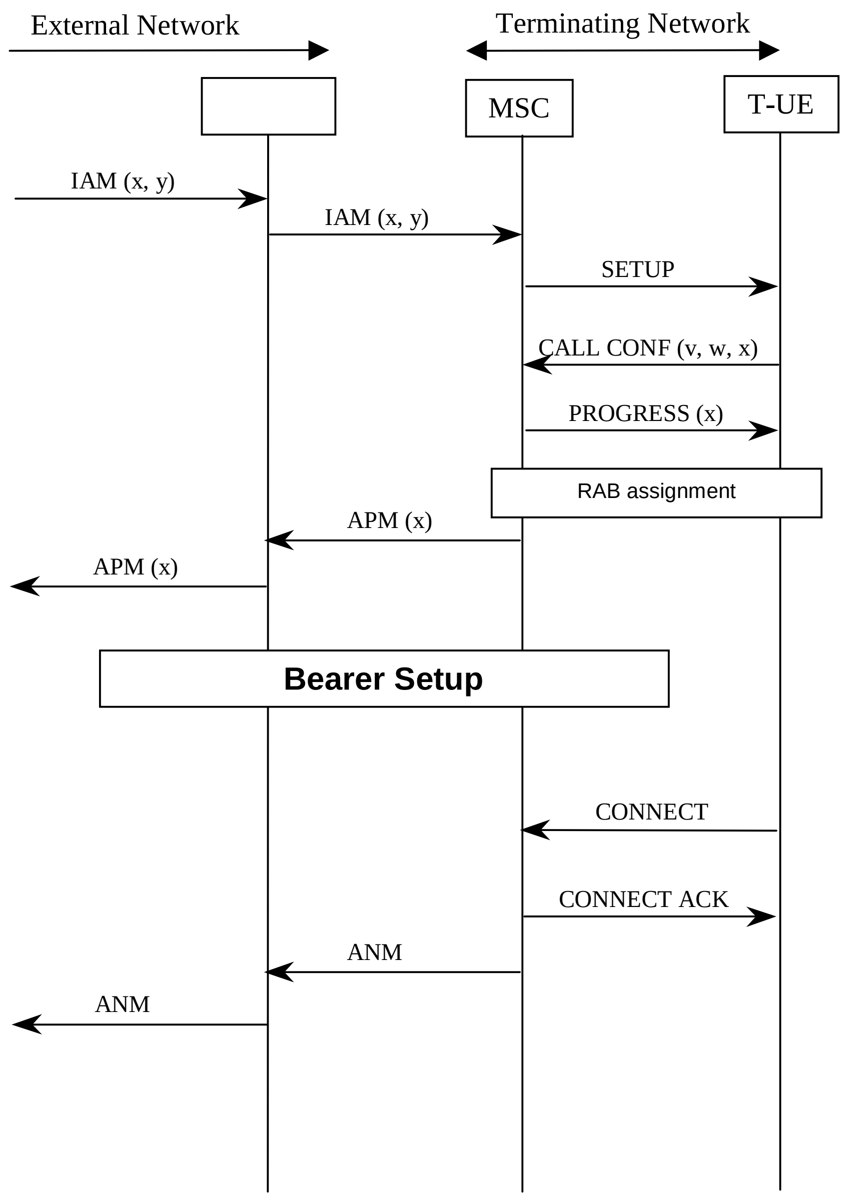

Figure 6.2.1/1 Successful MT call

- 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

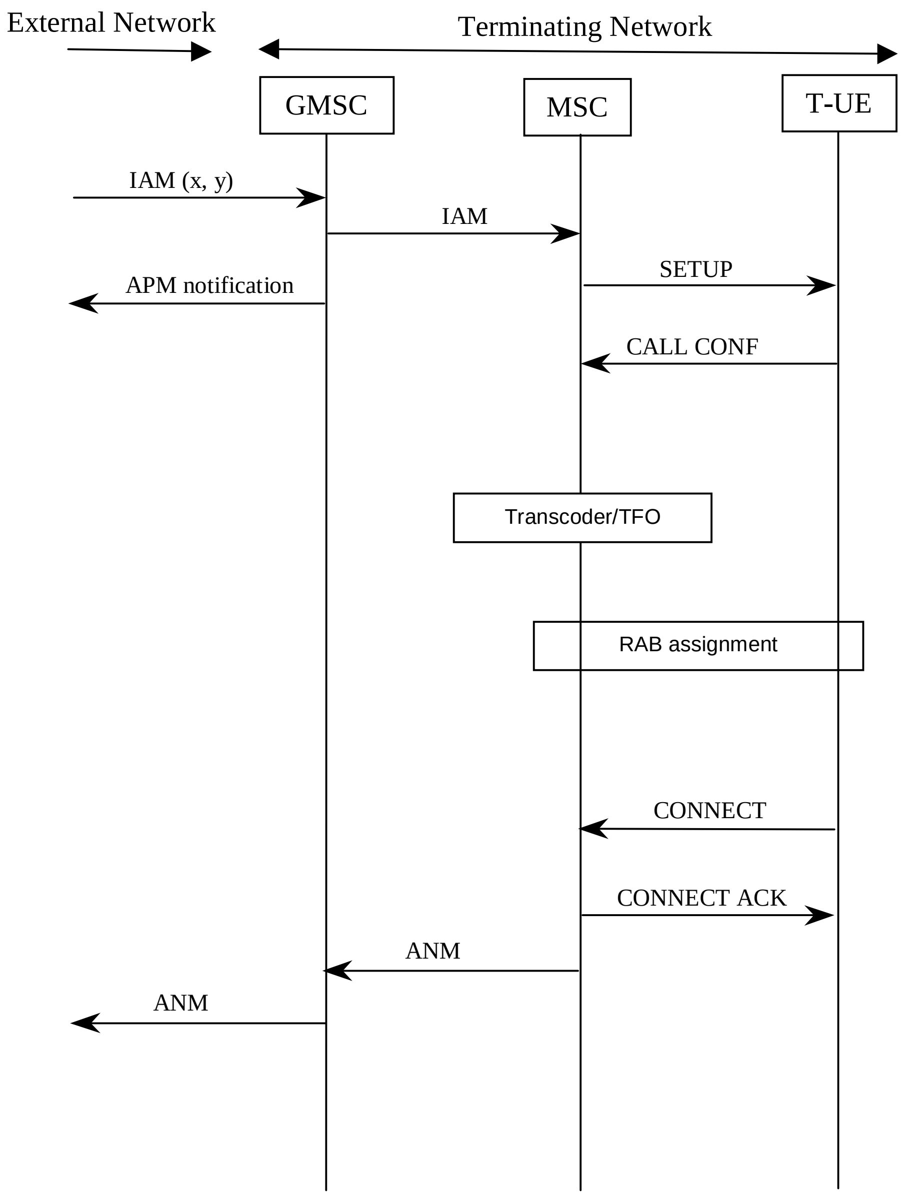

Figure 6.2.2/1 Unsuccessful MT call due to APM-user not supported

- 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

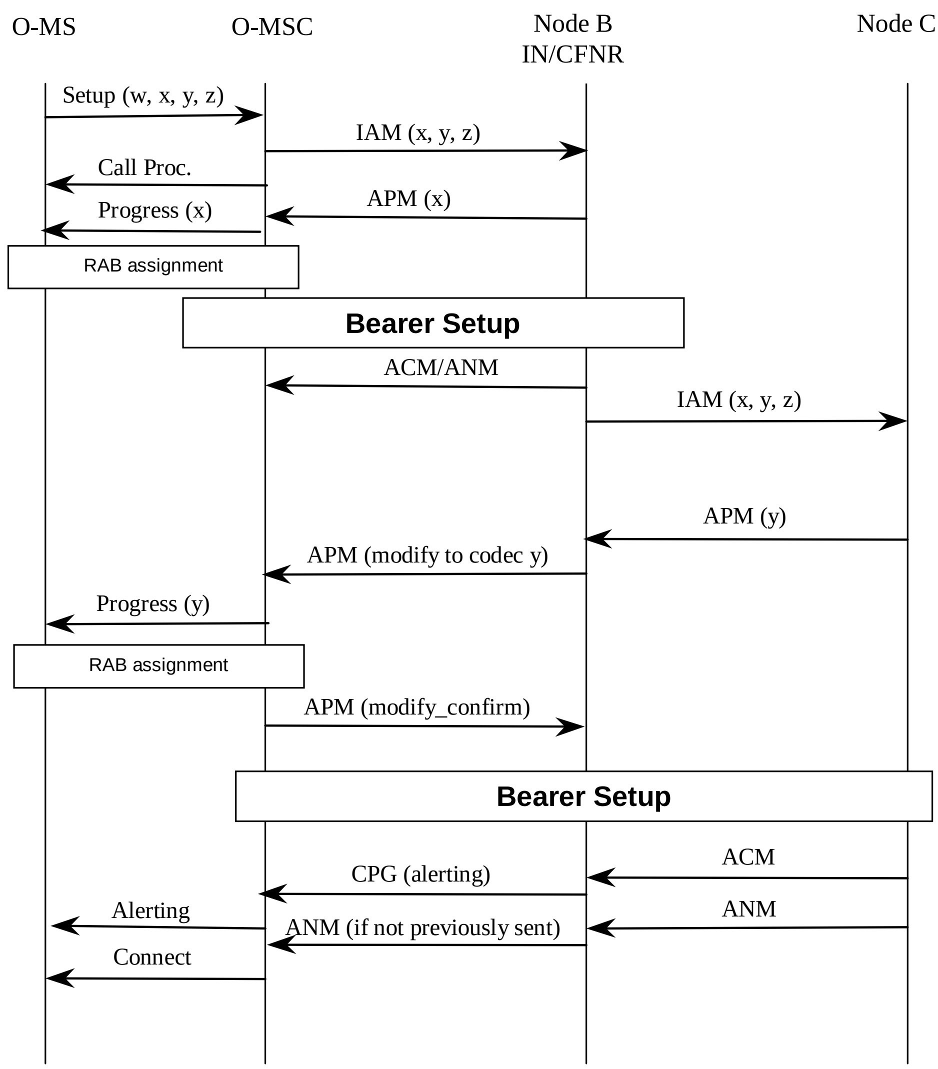

Figure 6.3/1 Interactions with IN or CFNR SS at call setup

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.

-

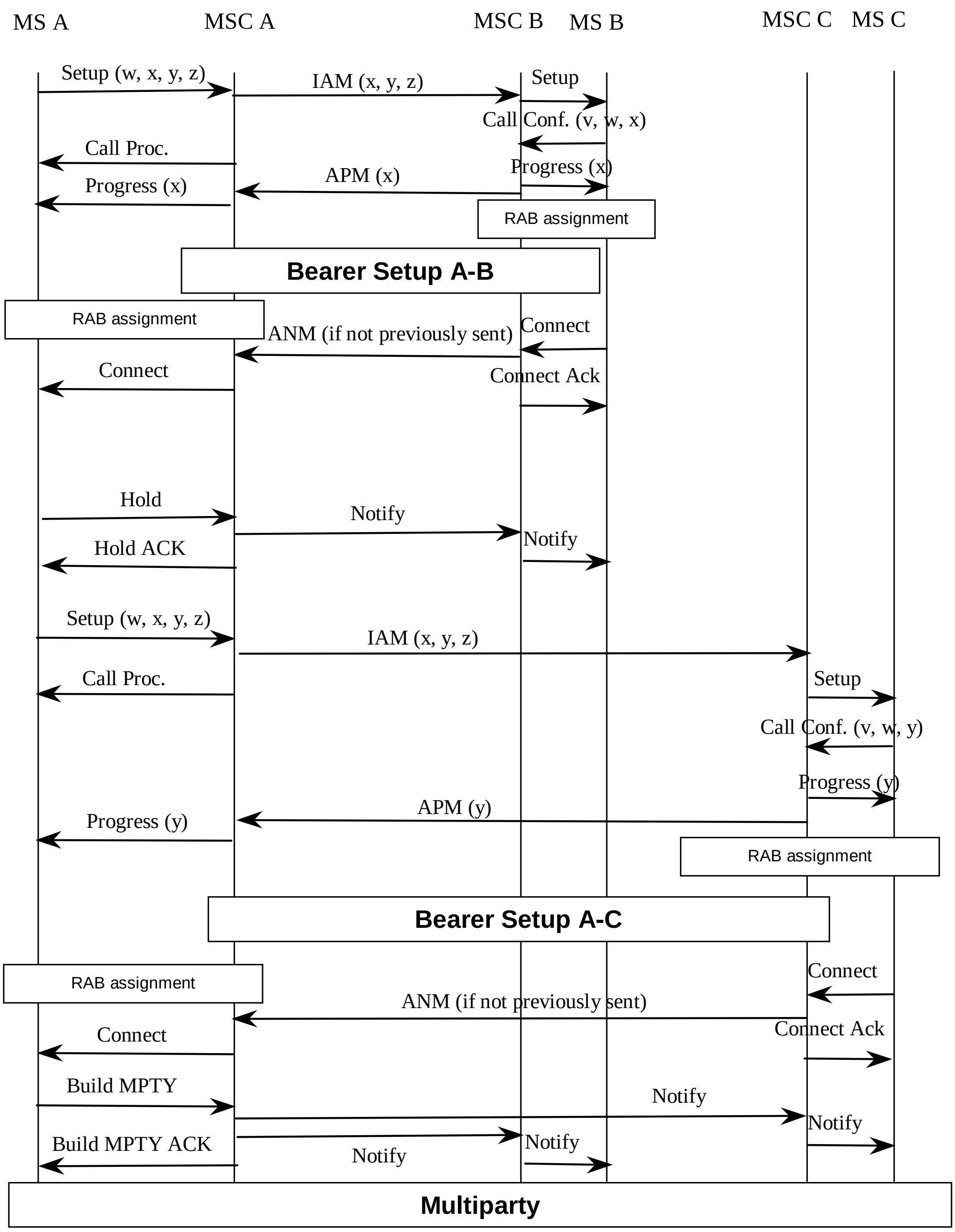

Figure 6.4/1 Interactions with Multiparty SS

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

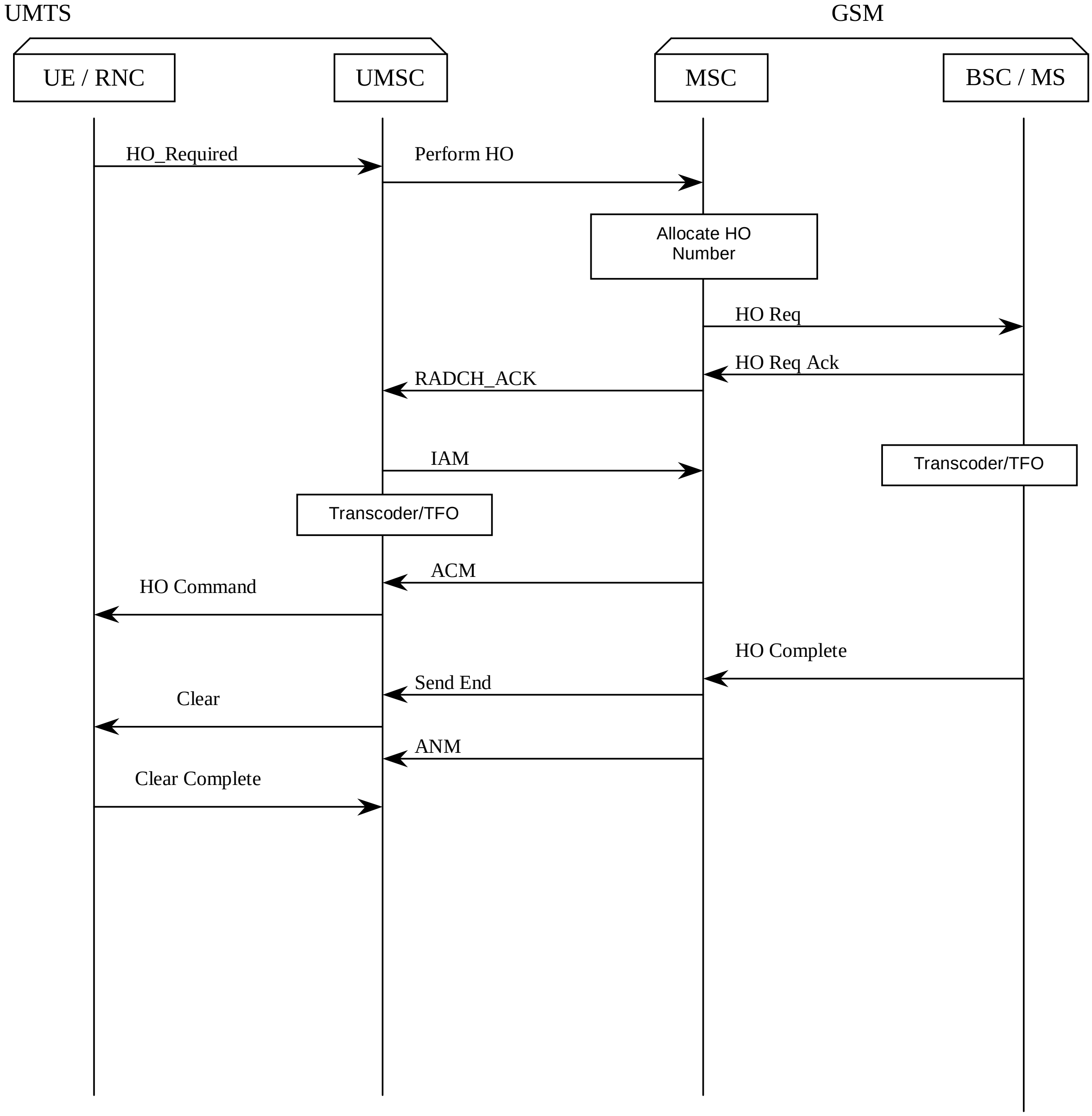

Figure 6.5/1 UMTS to GSM Handover after TrFO establishment

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 ab.cde

Status of Technical Specification ab.cde		
Date	**Version**	**Comments**
September 1999	0.0.0	First draft prepared by the rapporteur
October 1999	0.1.0	2^{nd} draft prepared by the rapporteur (Updated version from Abiko.)
November 1999	0.2.0	3^{rd} draft prepared by the rapporteur
Text and figures: **Stylesheet:** 3gpp_70.dot **Rapporteur**: Toshiyuki Tamura / Heinz-Peter Keutmann **Company:** NEC Corporation / Ericsson L. M.		

History

Document history		

3GPP TSG_N2#12
WG N2 Meeting, Phoenix, USA
15-19 November, 1999

T doc N2-99J60

LIAISON STATEMENT

Source: **LM Ericsson (Sweden) on behalf of 3GPP TSG-N2**

To: **ITU-T SG11 Q.12/11**

CC: **3GPP S4**

Title: **REFERENCES TO 3GPP CODEC INFORMATION**

Document for: **INFORMATION**

3GPP appreciated the work ongoing in ITU-T SG11 on the codec negotiation capabilities within the Bearer Independent Call Control (BICC) protocol. 3GPP has requested to have an organisation ID value assigned for 3GPP within the ITU-T draft recommendation Q.BICC. However, 3GPP noted that the Ottawa meeting of Q.12/11 requested a reference to the specification that describes the 3GPP codec information.

This specification is under the responsibility of 3GPP TSG-SA4 and is referred as TS 26.103. The intention is to approve this specification in December 1999.

TSG-S4 is currently planning to include the following information in the 3GPP TS 26.103:

- GSM Codec Types and Configuration parameters (FR, HR, EFR, AMR)
- UMTS Codec Types and Configuration parameters (AMR only at this point)
- Codec Compatibility Elements

3GPP would like to continue the cooperation between ITU-T SG11 and 3GPP and is looking forward to your recommendation Q.BICC

An example of the 3GPP reference is provided in Annex 1.

ANNEX 1 3GPP Reference

		Doc. Number	Version	Status	Issued Date	Location*
Specification[1] ARIB/TTC,CWTS,ETSI,T1,TTA		26.103	1.0.0	3GPP TSG SA WG4 approved	1999-12-03	http://www.arib.or.jp/IMT-2000/ARIB-spec http://www.etsi.org/smg/imtrefs/ http://www.t1.org/imtrefs http://www.tta.or.kr/imt2000/3gppspec
Standard[2]	ARIB/TTC					
	CWTS					
	ETSI					
	T1					
	TTA					

3GPP TSG CN2 **Tdoc 3GPP N2-99J86**
Phoenix, USA
15-19 November 1999

LIAISON STATEMENT

Source: **TSG CN WG2**

To: **TSG SA WG3**

Subject: **Confirmation of TS 33.102 section 6.3.4 Distribution of authentication vectors between VLRs**

Purpose: **For immediate action (before 18th of November 1999)**

TSG-N2 is making a CR to TS 29.002 (MAP) based on TS 33.102. In this work, we have a question on the TS 33.102. **N2 kindly asks S3 to answer to the questions by coming Thursday morning, 18 November 1999 (GMT -0700).**

In section 6.3.4, Figure 11 shows that SN/VLRn can send *Authentication data request* with IMUI to SN/VLRo. We could not understand the requirement and/or the service driver of using IMUI as a user identity in this case. Please tell us the requirement and/or the service driver.

The background of the question is the followings. The MAP_SEND_IDENTIFICATION request that corresponds to *Authentication data request* is used to retrieve the IMSI and authentication data by using TMSI, and the main purpose of the MAP_SEND_IDENTIFICATION service is to obtain the IMSI so far. Moreover, if the SN/VLR has already obtained the IMSI, it sends MAP_SEND_AUTHENTICATION_INFO request to the HLR in order to retrieve the authentication data.

Another issue is that N2 would like to raise is associated to the use of the limited radio interface resources. N2 understands that the feature of distribution of authentication vectors is initiated upon received request from the mobile for a location update. However, in the normal cases the mobile will identify itself with the TMSI. Indeed it is possible to request the IMSI from the mobile (which can be encrypted on the air interface, i.e. E-IMSI), but N2 would like to remind S3 that the radio resources are considered to be limited. N2's opinion is that unnecessary messages on the radio interface should be avoided.

Therefore, N2 would like to receive a clarification of the requirement that an IMSI can be available when the subscriber roams into a new VLR area and requires retrieval of authentication vectors. Normally, only the TMSI would be available.

3GPP TSG-CN WG2
Phoenix, USA, 15-19 November 1999

Document ***N2-99J78***

From: **TSG-N2**

To: **S2**

Cc: **TSG-N1**

Liaison Statement on the Introduction of Reserved Service Labels in the APN

N2 would like to inform S2 that it has reviewed and approved the attached change request (Tdoc N2-99J76) against TS 23.003 introducing Reserved Service Labels in the APN. N2 would like confirmation that the proposed changes meet the requirements to support Mobile IP, DCHP and other similar services.

3GPP/SMG Meeting #8
Phoenix USA 15-19 Nov 1999

***Document* N2-99J76**

e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

23.003	**CR**	**011r2**	Current Version:	3.2.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	CN#6	for approval	X	strategic	
list expected approval meeting # here ↑		for information		non-strategic	

(for SMG Use only)

Form: CR cover sheet, version 2 for 3GPP and SMG *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: *(at least one should be marked with an X)*	(U)SIM		ME		UTRAN / Radio		Core Network	X

Source:	Ericsson	**Date:**	1999-11-16
Subject:	Introduction of Reserved Service Labels in the APN		
Work item:	GPRS Phase 2		

Category:	F	Correction		**Release:**	Phase 2	
	A	Corresponds to a correction in an earlier release			Release 96	
(only one category	B	Addition of feature	X		Release 97	
shall be marked	C	Functional modification of feature			Release 98	
with an X)	D	Editorial modification			Release 99	X
					Release 00	

Reason for change: The PDP type IP has been extended to allow the separation of PDP context activation and ISP Environment setup. These extensions support e.g DHCP end-to-end and Mobile IP.
In order to help automatic APN selection, the concept of Reserved Service Label is introduced, which indicates that a special service is supported by the APN. The service offering is not exclusively coupled to the reserved APN: all APNs can support the new services if configured to do so by the operator.

Clauses affected: 9.1; 9.1.1

Other specs affected:	Other 3G core specifications	X	→ List of CRs:	23.060 CR 025
	Other GSM core specifications		→ List of CRs:	
	MS test specifications		→ List of CRs:	
	BSS test specifications		→ List of CRs:	
	O&M specifications		→ List of CRs:	

Other comments:

<--------- double-click here for help and instructions on how to create a CR.

9 Definition of Access Point Name

In the GPRS backbone, an Access Point Name (APN) is a reference to a GGSN. To support inter-PLMN roaming, the internal GPRS DNS functionality is used to translate the APN into the IP address of the GGSN.

9.1 Structure of APN

The APN is composed of two parts as follows:

- The APN Network Identifier which defines to which external network the GGSN is connected to and optionally a requested sevice by the MS. ~~service to be offered.~~ This part of the APN is mandatory.
- The APN Operator Identifier which defines in which PLMN GPRS backbone the GGSN is located. This part of the APN is optional.

The APN Operator Identifier is placed after the APN Network Identifier. An APN consisting of both the Network Identifier and Operator Identifier corresponds to a DNS name of a GGSN and has a maximum length of 100 octets.

The syntax of the APN shall follow the Name Syntax defined in RFC 2181 [14] and RFC 1035 [15]. The APN consists of one or more labels. Each label is coded as one octet length field followed by that number of octets coded as 8 bit ASCII characters. Following RFC 1035 [15] the labels should consist only of the alphabetic characters (A-Z and a-z), digits (0-9) and the dash (-). The case of alphabetic characters is not significant. The APN is not terminated by a length byte of zero.

NOTE: A length byte of zero is added by the SGSN at the end of the APN before interrogating a DNS server.

For the purpose of presentation, an APN is usually displayed as a string in which the labels are separated by dots (e.g. “Label1.Label2.Label3”).

9.1.1 Format of APN Network Identifier

The APN Network Identifier shall contain at least one label and shall have a maximum length of 63 octets. An APN Network Identifier shall not start with the strings “rac”, “lac” or “sgsn” and it shall not end in “.gprs”. It shall also not take the value “*”.

In order to guarantee uniqueness of APN Network Identifier within the GPRS PLMN(s), an APN Network Identifier, ~~without considering a possible starting Reserved Service Label,~~ containing more than one label corresponds to an Internet domain name. This name should only be allocated by the PLMN to an organisation that has officially reserved this name in the Internet domain. Other types of APN Network Identifiers are not guaranteed to be unique within the GPRS PLMN(s).

An APN Network Identifier consisting of 3 or more labels and starting with a Reserved Service Label, or an APN Network Identifier consisting of a Reserved Service Label alone, shall indicate, that for this APN, the GGSN supports additional services. Reserved Service Labels and the corresponding services they stand for are to be agreed among operators.

3GPP TSG-N WG2 **Tdoc N2-99J84**
Phoenix (AZ), US
15 –19 November 1999

From: **TSG N WG 2**
To: **TSG SA WG1, TSG SA WG2**

LS on Open Issues on 3G to 2G Handover (packet Switched domain)

N2 would like to ask S1 and S2 advice on how to handle the UMTS PS domain to GPRS handover from a service aspects point of view, in view of the following information we provide below.

In UMTS a new version of the GTP protocol will be used (GTPv1). GTPv1 enables 3G service features that are not provided by GTPv0, such as multiple levels of QoS for a single PDP address, and packet filtering rules, as well as RSVP and differentiated services support. It is in order evaluating what is the impact of these different service features on the support of 3G to 2G handover.

If a user equipped with a dual mode terminal is in 3G coverage he/she may act so that multiple PDP contexts are activated (for instance to support multimedia). As he/she moves to a 2G coverage area, under the domain of a 2G SGSN, then it is not possible to keep all but one of the PDP contexts associated to a single PDP address. Also, the user has no possibility to modify any filtering rule that had been set up using GTPv1 procedures.

It's not nice to make an arbitrary selection of which PDP context is to be preserved, since the importance of each PDP context to the user is not under the control of the network. However, some standard on which PDP context to be kept and with which QoS must be defined for 3G to 2G handover.

N2 urges S1 and S2 to provide advice on which approach to take in order to provide consistent service provision while allowing for 3G to 2G handover, since this may have impact on stage 3 specifications for Release '99.

Title: **LS on introduction of Follow Me for the transferred Chapter 4 of 29.002**

Source: **TSG N2**

Liaison to: **S2**

Since the chapter 4 of TSG 29.002 is transferred to S2, N2 wants to inform S2 about changes which are requested by introducing the feature Follow Me (GSM 02.94, 03.94) in this chapters. N2 kindly asks S2 to prepare the changes. The revised Tdoc at N2 for the changes at MAP is N2-99J88.

For information the paragraphs below are showing the proposed changes related to the old chapter 4 of MAP specifications.

4.4.2 Interface between the HLR and the gsmSCF (J-interface)

This interface is used by the gsmSCF to request information from the HLR (via the Any-time Interrogation function) or to allow call independent related network- or user-initiated interaction between an MS and the gsmSCF (via the USSD function). Support of the gsmSCF-HLR interface is a network operator option. As a network operator option, the HLR may refuse to provide the information requested by the gsmSCF.

4.4.2A Interface between the HLR and the HLR (?-interface)

This interface is used to allow call independent user-initiated interaction between an MS and an HLRb (which is different from the initiating subscriber's HLRa) via the USSD function. Support of the HLRa-HLRb interface is a network operator option.

3GPP TSG_CN2#12 **Tdoc N2-99K04**
WG N2 Meeting, Phoenix, USA,
15-19 November 1999

Source: TSG N2

To: 3GPP TSG SA-2

CC: 3GPP TSG CN

Title: LS to S2 on establishment of a QoS taskforce

Agenda item: 9.2

Document for: Approval

TSG N2 would like to thank S2 for their presentation of the 3G TS 23.107 on "QoS Concept and Architecture". The presentation of the 3G TS23.107 raised concern on the specification of the required protocol support since the impacts of QoS on the stage 2 specifications is still ongoing (e.g. 3G TS 23.060).

N2 noted that QoS support is an essential part of the Release 99 package and noted that only two meetings are left to the QoS specification in the core network protocol specifications (e.g. 3G 29.060 and 29.002). Therefore, N2 initiated a taskforce to progress the work on the introduction of the QoS concepts within the protocol specifications.

N2 also realised that QoS stage 2 expertise is required to ensure that the QoS concepts can be introduced in the protocol specification in time for Release 99. Therefore, participation of S2 (i.e. S2 – QoS Adhoc) is essential to progress the activity in this Taskforce.

N2 would like to create a taskforce on QoS with the experts from N2 and S2 to complete the work on the introduction of the QoS concepts within the required 3G TS specifications. Mr. Alessio Casati (Lucent Technologies), acasati@lucent.com kindly volunteered to act as convenor from N2 perspective. N2 would appreciate if S2 will also assign a convenor from S2 perspective.

N2 would appreciate a response of S2 on their email list (i.e. 3GPP_TSG_CN_WG2@3gpp.org) before 7th of December 1999 to ensure proper presentation of the status of the taskforce to the coming TSG CN plenary #06.

3GPP TSG CN WG2 **Tdoc N2-99K66**
Phoenix, AZ, USA
15-19 November 1999

Source : TSG N2

To: TSG S2
CC: TSG S1

Title: Liaison Statement to S2 on Combined Mobility Management

CN2 would like to inform S2 on the outcome of their discussion on Combined Mobility Management. It was concluded that this WI could not be finalized for R99 and that the work is proposed to be transferred to Release 2000 for completion.
The issues addressed by S2 in their LS (document S2-99947) will be taken into consideration.

3GPP TSG-SA Codec Working Group *TSGS4#6(99)267*
TSG-S4#6: September 8-10, 1999, Sophia-Antipolis, France

Liaison To:	**TSG-N2**
From:	**TSG-S4 (Codec Working Group)**
cc:	**S2, R3, TSG-CN**
Subject:	**Response to N2 LS on Tandem Free and Out of Band Transcoder Control**

1. Introduction

S4 welcome the activities aiming through Out-of-Band means at enabling Tandem Free Calls. This will enable Tandem Free Operation from Call Set-up to Call Release in a very clean way.

S4 has looked and analyzed the N2 Liaison Statement (N2-99976) and especially its attached technical report.

The rest of this document contains comments on the mentioned technical report as well as some issues that S4 believe should be considered by N2 when developing the specifications related to the Out-of-Band control of the Transcoders.

S4 would appreciate if N2 could keep them informed on the progress of N2 activities on the control of the Transcoders.

2. Proposed Definitions

In order to avoid future misunderstanding and clarify the content of this and future liaisons, S4 would like to propose the following definitions for Tandem Free related items. If acceptable, S4 will propose to TSG-SA to include these definitions in the relevant 3GPP vocabulary document:

Transcoder (TC): Physical device present in the network responsible for the transcoding of the speech data between two speech codecs or coding schemes (The Transcoder may also include other functions, i.e. Rate Adaptation in GSM).

Tandem Free Operation (TFO): Configuration of a Speech or Multimedia call for which Transcoders are physically present in the communication path but transcoding functions are disabled or partially disabled. The Transcoders may perform control and/or protocol conversion functions.

In-Band Transcoder Control: Capability for a system to control the operation mode of a Transcoder through in-band signaling, embedded in the speech frame. For example, in GSM, the speech codec in use and DTX parameters are provided in-band by the BTS to the transcoder, on a call per call basis, to control the operation mode of the transcoder.

In-Band TFO Protocol or In-Band TFO: Inter-Transcoder protocol first standardized for the GSM system, allowing the Transcoders to recognize the tandem free capability of the remote end, to identify a potential codec mismatch and establish when possible Tandem Free Operation. The protocol messages are carried through bit-stealing before TFO establishment and embedded in the synchronization pattern after TFO establishment.

Transcoder Free Operation (TrFO): Configuration of a Speech or Multimedia call for which Transcoders are not present in the communication path.

Out-of-Band Transcoder Control: Capability of a system to control the operation mode of a Transcoder on a call per call basis through out-of-band signaling. The operation mode should at least include the codec type and associated parameters (DTX activation, allowed set of codec modes when a Multi-Rate codec is used…). Out-of-Band Transcoder Control is required to establish Transcoder Free Operation (TrFO).

Out-of-Band TFO: Possibility to establish Tandem Free Operation through Out-of-band signaling. We could envisage establishing TFO through out-of band signaling but leave the Transcoder in the communication path to monitor external events and revert to a normal configuration if required. If this option is kept, the out-of-band control protocol will end up by either bypassing the transcoders or including the transcoders without activating the transcoding functions.

3. Comments on the Out-of-Band / In-Band Comparison

The comments in this section must be considered as additional information to sections 7 and 8 of the technical report 3G TR ab.cde produced by N2.

Service Limitation

In-Band TFO was developed for speech services only as required by the TFO stage 1 document (see GSM 02.53). The protocol could also be extended to services requiring more than 64 kbps transmission links. This need has not been expressed so far. S4 still recognize that TFO was first developed for speech telephony services and not for Multimedia calls. Nevertheless it should be noted that In-Band signaling has been used in Multimedia for a long time. In ITU-T H.324 and H.323, transcoding functions are avoided using the capabilities offered by the [in-band] H.245 control channel.

The transmission saving capability (i.e. 8 or 16 kbps) has been taken into account in In-band TFO (see GSM 08.62) and a specific annex (B) has been dedicated to this.

In-Band TFO assumes 64 kbps links, but since it is In-Band it is independent of the transport network assuming that this Transport Network will either transport the 64 kbps or if used for transmission saving too, will transport the 8 or 16 kbps stream.

In-Band TFO will be compatible with GSM but is likely to be also adopted by other 2G and 3G systems (i.e. IS-136, IS-95, cdma2000).

Configuration

The Hardware configuration issue is partly dependent on implementation aspects and saving can be achieved in configurations processing a high number of calls as expected in 3G systems. The monitoring of the synchronization is required independently of the fact that Out-of-Band or In-Band signaling has being used for establishing TFO. In any case, synchronization monitoring is required somewhere in the radio network or BSS for either Tandem Free Operation or Transcoder Free Operation.

Quality of Service

For in-band TFO, the impact on speech quality has been extensively evaluated through formal and informal listening tests carried out under SMG11 responsibility. No degradation was found.

The bit-stealing is done on speech calls only and not for data calls. TFO establishment starts after the circuit is opened and therefore it is known then whether it is a speech or data call.

Maintenance for future expansion

When a new codec type is introduced the existing transcoders do not need to be upgraded, in the sense that if they’re not compatible this will be detected by the entity in charge of the codec mismatch resolution. If the transcoder is not upgraded the new codec type will be considered unknown and TFO not established. Alternatively, the network could decide to change transcoders to allow TFO establishment. In most cases, the core network equipment will be made of Pools of Transcoders with different capabilities.

The transit nodes can indicate what is their Preferred Codec in the current GSM TFO and this will be the case also for 3G, therefore the relocation can be done taking into account this need.

4. Potential Issues

This section lists a number of remarks and issues identified in the TR produced by N2:

- The report focuses on the signaling required to establish a Transcoder Free configuration. S4 believe that an out-of-band signaling solution should also offer the possibility to establish a Tandem Free Operation where a transcoder is inserted in the communication path but the transcoding functions are bypassed. This solution could be beneficial when TFO must be discontinued during the call (DTMF, Multiparty call activated…). In that case, the transcoder would only have to monitor the synchronization and revert to normal transcoding if commanded by the Core Network or when TFO is discontinued.

- The 3G TR ab.cde does not address the synchronization monitoring. How will that be performed in Transcoder Free Operation?
- In case of Transcoder Free Operation, which format will be used to convey the speech data? What is the compatibility with GSM?
- How could Transcoder Free Operation be implemented on an existing GSM network, providing that the TRAU is part of the BSS? or should the Core Network be equipped with internal Transcoders in that case?
- How will the management of different AMR configurations (i.e. Active Codec Modes, Supported Codec Modes, Maximum number of codec modes) be done? What is the impact on the BICC specification?
- How will the in-band signaling used in GSM for the AMR link adaptation (see GSM 05.09) be treated in case of Transcoder Free Operation between GSM and 3G?
- As for In-Band TFO, downlink DTMF will break Transcoder Free Operation and will be subject to the non-ideal transparency performances of the speech codec. S4 believe that out-of-band Layer 3 signaling for downlink DTMF should be considered.

5. Conclusion

S4 understand that certain aspects of the Physical Layer for the speech services can have some impact on TFO or TrFO independently of the Out-of-Band or In-Band approach. We feel that it is important that N2 and S4 collaborate on these aspects, especially since most of them were assessed and taken into account in the context of the In-Band TFO work carried out for GSM[1]. S4 consider that the complexity of the In-band TFO protocol essentially comes from the flexibility and constraints introduced on the physical layer of the Air Interface; Maximum of 4 modes in the AMR Active Codec Set, freedom for network manufacturers to support any set of AMR codec modes, alternating transmission of Codec Mode Requests and Codec Mode Indications. It is still unclear if the UTRAN will not have similar limitations. An out-of-band protocol for TFO or TrFO will also have to consider these constraints, for 3G-3G or 2G-3G interoperability.

Furthermore we understand that TFO and TrFO may not be transparent to the RAN and we may have to involve TSG-RAN (R2 and R3) and S2.

[1] Note that the maintenance of the GSM In-Band TFO specifications is being transferred from SMG11 to S4.