3GPP TSG CN Plenary Meeting #14 **NP-010563**
Kyoto, Japan. 12th - 14th December 2001.

Source: **MCC, CN3**

Title: **LS PACK – of all LSs sent from CN3 since NP#13**

Agenda item: **6.3.1**

Document for: **INFORMATION**

The following Liaison Statements were agreed and sent by CN3:

Tdoc #	Tdoc Title	LS to	LS cc	Attach.
N3-010446	SIP Signalling and CODEC Issues	GERAN, SA2, SA4	CN1	N3-010390
N3-010481	PDP context based Go Interface	SA2	none	none
N3-010483	Signalling Transparency [Re. OSV-01043 and S2-012321]	GERAN, SA2	CN1	N3-010387, N3-01088
N3-010589	Response to LS on data rates for CS data services in UTRAN	GERAN WG1	RAN3, SA1	N3-010583
N3-010610	QoS Mapping for IMS to and CC:	SA4	SA2, RAN2	N3-010530
N3-010602	Addition of the H.324 M codec to TS 26.103	SA4		

3GPP TSG CN WG3 Meeting #19 **N3-010446**
Brighton, U.K. 15th - 19th October 2001

Title:	**Reply Liaison Statement on SIP Signalling and Codec Issues**
Source:	**CN3**
To:	**GERAN, SA2 and SA4**
Cc:	**CN1**

Response to: LS (N1_011334/N3-010390) on SIP Signalling and Codec Issues from CN1

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Attachments: **N3-010390** [original LS to CN3 from CN1]

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1. Overall Description:

This LS is presented for information to GERAN, SA2 and CN1. SA4 is kindly asked to provide the requested information to GERAN and SA2.

In the LS N1-011076 on SIP Signalling and Codec Issues from the Joint TSG- GERAN/SA2 meeting to CN1, the following question was raised:
"If AMR is used is there a mechanisms that can enforce the use of an AMR mode that can be carried on a physical HR channel (i.e. AMR 795 or lower) within the RTP for carrying Optimised Voice in GERAN ?"
CN1 passed this question to CN3 and SA4.

CN3 would like to inform GERAN and SA2 that the transport of AMR within RTP was defined by SA4 in TS 26.234 and 26.235. For this reason, CN3 feels that SA4 is in the best position to answer the above question.

2. Actions:

To SA4.

ACTION: CN3 kindly asks SA4 to answer the following question raised by GERAN and SA2.

"If AMR is used is there a mechanisms that can enforce the use of an AMR mode that can be carried on a physical HR channel (i.e. AMR 795 or lower) within the RTP for carrying Optimised Voice in GERAN ?"

3. Date of Next CN3 Meetings:

CN3#20	26th – 30th November 2001	Cancun, Mexico.

3GPP TSG CN WG3 Meeting #19 **N3-010481**
Brighton, U.K. 15th - 19th October 2001

Title:	**Liaison Statement on PDP context based Go Interface**
Source:	**CN3**
To:	**SA2**
Cc:	
Response to:	

Contact Person:

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Attachments: None

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1. Overall Description:

CN3 is currently standardizing the Go interface, based on the requirements in TS 23.207 by SA2. CN3 has found a possible cause of complications and would like to request for clarification/help from SA2.

When the GGSN receives a Create_PDP_Context_Request message including binding information, the GGSN requests an authorization decision from the PCF. The Go interface is currently specified to be IP flow based in TS 23.207. This means that the decision sent by the PCF is IP flow based. The PDP context may be used to serve many IP flows meaning that the GGSN shall combine the IP flow based policy information from the PCF decision to form policy information per PDP context.

CN3 sees that complex situations may occur if the GGSN receives contradicting policy information for the IP flows carried within one PDP context. Moreover, the GGSN needs to combine the authorized QoS information, the DiffServ code points and packet classifiers related to the IP flows carried by the PDP context.

CN3 feels that the IP flow based response to a PDP context based request complicates CN3's standardization work on the Go interface and the operations of the GGSN.

2. Actions:

To SA2 group.

ACTION: CN3 asks SA2 group to consider whether there are clear reasons to standardize the Go interface to be IP flow based, or whether the Go interface can be PDP context based.

3. Date of Next CN3 Meetings:

CN3_20 26th – 30th November 2001 Cancun, Mexico.

3GPP TSG CN WG3 Meeting #19 **N3-010483**
Brighton, U.K. 15th - 19th October 2001

Title:	**Liaison Statement on Signalling Transparency**
Source:	**CN3**
To:	**GERAN and SA2**
Cc:	**CN1**

Response to: LS (**OSV-01043/N3-010388**) on Signalling Transparency from S2/GERAN Joint Meeting on IMS and Optimised Voice
and Reply LS (**S2-012321/N3-010387**) on Signalling Transparency from SA2.

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Attachments: **N3-010387, N3-01088.** [Original LSs from S2/GERAN and S2]

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1. Overall Description:

This LS is to presented for information to SA2, GERAN and CN1.

In its LS OSV-01043, the SA2/GERAN Joint Meeting on IMS and Optimized Voice requested information on the current work status in CN3 for the „ case of a IMS user in a communication exchange to a non SIP user where a signaling translator is needed on the control plane to translate SIP messages to the call control used by the other party.“
Furthermore, this meeting assumed „that the control plane signaling transition is transparent to the end systems“, and requested a confirmation of this assumption from CN3.

In the response LS S2-012321, SA2 provided architectural guidelines and asked CN3 to take them into account. SA2 also confirmed the above assumption from LS OSV-01043.

CN3 would like to thank SA2 for their guidelines and would like to confirm that the current work in CN3 takes these guidelines into account.

CN3 is pleased to present the requested information to the involved groups:

- CN3 is specifying the interworking between the IMS and PSTNs (using either ISUP or BICC) in TS 29.163. It is the intention of CN3 to keep this interworking transparent to the end system.
- CN3 is also specifying the interworking between the IMS and external IP networks in TS 29.162. In Rel.5, the related work will be restricted to user plane transcoding and possibly to the interworking between SIP with the extensions of the 3GPP profile, as defined in TS 24.229, and standard SIP, as defined in RFC2543. It is the current working assumption in CN3 than such an interworking will be required and that it will be transparent to the UE.
  A joint meeting of CN1 and CN3 decided that CN1 investigates further, whether SIP with extensions of the 3GPP IMS profile and standard SIP require an interworking function in order to interoperate.
  CN3 decided not to specify the interworking to H.323 from IMS specification. Such an interworking can be provided by network entities outside the 3GPP IM CN subsystem.

3. Date of Next CN3 Meetings:

CN3_20 26^{th} – 30^{th} November 2001 Cancun, Mexico.

3GPP TSG CN WG 3 #20 **N3-010589**
Cancun, Mexico, 26-30 November 2001 **Agenda Item: 6**

Title:	**LS on data rates for CS data services in UTRAN**
Source:	**TSG CN WG3**
To:	**TSG GERAN WG1**
cc:	**TSG RAN WG3, TSG SA WG1**

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1. Overall Description:

CN WG3 thanks GERAN WG1 for their liaison in **[GP-012792]** relating to data rates for CS data services in UTRAN.

The specification in which the data rates are currently described when the access network is connected to the core network via the Iu-cs is ***3GPP TS 22.002 "Circuit Bearer Services (BS) supported by a Public Land Mobile Network (PLMN)"***. This specification is under the reponsibility of TSG SA WG1.

The services defined for UTRAN / UMTS in TS 22.002 are applicable only if the access network is connected to the core network via the Iu cs interface.

Services defined for GERAN / GSM in TS 22.002 are applicable only if the access network is connected to the core network via the A interface.

2. Actions:

NONE

3. Date of Next CN WG3 Meeting:

TSG CN WG3 #21 28th Jan. – 1st Feb 2002

4. Attachments:

N3-010583 / GP-012792. [original LS from GERAN1].

3GPP TSG CN WG3 Meeting 20 ***N3-010602***
Cancun, Mexico, 26th - 30th November 2001

Title:	**Liaison Statement on The addition of the H.324 M codec to TS 26.103**
Source:	**CN3**
To:	**SA4**
Cc:	
Response to:	-

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Attachments: none

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1. Overall Description:

CN1 and CN3 progress with the work item "Service change and UDI fallback" for Release 5. The solution chosen is based on Out-of-Band Transcoder Control (OoBTC) procedures defined in TS 23.153 (based on BICC CS2) for codec negotiation, selected codec modification, and list of available codecs modification.
BICC CS2 defines a mechanism for negotiating codecs at call set-up. This mechanism is described in Q.1902.4, chapter 8.3, "codec negotiation". It also supports a mechanism for modifying the active codec during a call, "codec modification", and re-negotiating the list of supported codecs during a call, "mid-call codec negotiation", which are described in Q.1902.4, chapter 10.4. The adaptation of this mechanism to 3GPP networks, Out-of-Band Transcoder Control, is described in TS 23.153.
The support of a service allowing both UDI multimedia and speech requires specific needs from the network :

- modification of the network bearer from a speech bearer to a 64 kb/s unrestricted bearer and vice-versa, during the active state of the call.
- when not possible, reserving a permanent 64 kb/s unrestricted bearer for the whole duration of the call, even when the call starts in speech mode. Unfortunately, the combination of parameters needed to achieve this is likely to be rejected by non-supporting transit networks, leading to unwanted call rejections.
- notification of a request to service change from one side to the other, and reception of the result (acknowledgement or rejection) from the other party or from the transit network in-between. Unless such a mechanism is available, both sides have to synchronise by other means their intention to change the service (verbal agreement beforehand), which is prone to errors and may lead to unsatisfied customers.
- a network-assisted service change (where only one user needs to initiate the service change), both to avoid an unwanted verbal synchronisation and to avoid relying on the clever behaviour of the users to recover from a possible failure (if either one of the terminals or the networks does not support the modification).

2. Actions:

To SA 4 group.

To be able to finalise the WI, it is required that SA 4 includes the H.324M codec in TS 26.103, so make it possible to use H.324 M in the mentioned procedure.

3. Date of Next CN3 Meetings:

CN3#21 28th Jan - 1st Feb 2002 Sophia Antipolis, France.

3GPP TSG CN WG3 Meeting #20 **N3-010610**
Cancun, Mexico. 26th - 30th November 2001.

Title:	**LS on "Mapping of SDP parameters in UMTS QoS parameter".**
Source:	**CN3**
To:	**SA4**
Cc:	**SA2, RAN2**

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Attachments: **N3-010530.** [Discussion document]

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1. Overall Description:

The ongoing work in CN3 has identified the need to specify the mapping from SDP parameters describing a multimedia session into UMTS QoS parameters for conversational service in IMS. The UMTS QoS parameters is defined in TS 23.107 and in section 10.5.6.5 of the TS 24.008.

CN3 is progressing the work on defining the above mappings for 3GPP codecs including AMR-NB. CN3 discussed the concept attached in the tdoc N3-010530 as mapping from SDP parameters into the UMTS QoS parameters for AMR-NB.

2. Actions:

To SA4 group. CN3 asks SA4 group to provide:-

- Comments on the discussed mapping from SDP parameter into the UMTS QoS parameters for AMR-NB.
- Guidance on the mapping from SDP parameter to UMTS QoS parameter for other 3GPP codecs (e.g. H.263, AMR-WB …)
- General guidance on the mapping from SDP parameter to UMTS QoS parameter for unknown codecs.

3. Date of Next CN3 Meetings:

CN3_21 28^{th} Jan – 1^{st} Feb 2002 Sophia Antipolis, France.