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Contents

Foreword

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y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

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Introduction

1 Scope

The present document investigates architectural requirements and architectural alternatives for using existing CS bearers in association with an IM session. The document considers overall requirements, architectural requirements, considerations for evaluation of potential architectural solutions and proposed alternative architectures.

The Feasibility Study covers different solutions for offering existing IMS simultaneous services (real-time media + non-real-time media) also in GERAN, where conversational PS spectrum efficiency is too low.

The target is to seek for an architectural solution that is completely transparent for the end-user, and is easily interoperable with existing IMS services&networks that don't use this solution.

The solutions studied within the Feasibility Study are not necessarily restricted by existing service requirements. However, if an alternative is chosen to be included in specifications, it must be cross-checked with existing service requirements.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

[1] 3GPP TS 22.228, "IP Multimedia Subsystem; Stage 1"

[2] 3GPP TS 23.228, "IP Multimedia Subsystem (IMS); Stage 2"

[3] RFC 3725 "Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)"

[4] RFC 3261 "SIP: Session Initiation Protocol"

[5] RFC 3264 "An Offer/Answer Model with the Session Description Protocol (SDP)"

[6] 3GPP TS 23.141 "Presence Service; Architecture and functional description"

[7] 3GPP TS 23.087 "User-to-User Signalling (UUS) supplementary service; Stage 2"

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply.

CS/CSB: an approach to combine CS calls and IMS sessions, where

- the session and CS speech bearer control is kept on the CS domain (as in TS 24.008)
- the session control for IMS services is done via SIP/IMS

IMS/CSB: an approach to combine CS calls with IMS sessions, where

- the session control for IMS services (including speech) is done via SIP/IMS using a logic residing in the IMS (AS)

- the CS speech bearer control is done via CS signalling (as in TS 24.008)

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

Editor's Note: abbreviations which are not used will need to be deleted.

3pcc	Third Party Call Control
AS	Application Server
BG	Border Gateway
BGCF	Breakout Gateway Control Function
CBCF	Circuit Bearer Control Function
CBOF	Circuit Bearer Originating Function
CBTF	Circuit Bearer Terminating Function
CN	Core Network
CS	Circuit Switched
CSB	Circuit Switched Bearer
CSCF	Call Session Control Function
DTM	Dual Transfer Mode
GGSN	Gateway GPRS Support Node
GERAN	GSM/EDGE Radio Access Network
HSS	Home Subscriber Server
I-CSCF	Interrogating-CSCF
IETF	Internet Engineering Task Force
IM	IP Multimedia
IM CN SS	IP Multimedia Core Network Subsystem
IMS	IP Multimedia Core Network Subsystem
IMS ALG	IMS Application Level Gateway
IMSI	International Mobile Subscriber Identifier
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
IP-CAN	IP-Connectivity Access Network
ISUP	ISDN User Part
MAP	Mobile Application Part
MGCF	Media Gateway Control Function
MGF	Media Gateway Function
MGW	Media Gateway
NAI	Network Access Identifier
NA(P)T-PT	Network Address (Port-Multiplexing) Translation-Protocol Translation
OSA	Open Services Architecture
P-CSCF	Proxy-CSCF
PDF	Policy Decision Function
PDN	Packet Data Network
PDP	Packet Data Protocol e.g., IP
PEF	Policy Enforcement Function
PLMN	Public Land Mobile Network
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RAB	Radio Access Bearer
RAN	Radio Access Network
S-CSCF	Serving-CSCF
SDP	Session Description Protocol
SGSN	Serving GPRS Support Node
SLF	Subscription Locator Function
SSF	Service Switching Function
SS7	Signalling System 7
SIM	Subscriber Identity Module

SIP	Session Initiation Protocol
SGW	Signalling Gateway
UE	User Equipment
UMTS	Universal Mobile Telecommunications System
URL	Universal Resource Locator
USIM	UMTS SIM
UTRAN	UMTS Terrestrial Radio Access Network
UUS	User-to-User Signalling
VoIP	Voice over IP

4 Overall Requirements

Editor's Note: This section will describe the overall requirements from a user/network operator point of view

5 Architectural requirements and considerations

Editor's Note: This section will describe requirements that apply to the architecture design and considerations which will be used when making decisions on the preferred architectural alternative.

5.1 Basic assumptions

- Although the scope is mainly targeted at GERAN, the solution is (at least technically) assumed to be applicable to GERAN and UTRAN.
- A CSB UE requires DTM capability (in case of GERAN access) and MultiRAB capability (in case of UTRAN access) - see subclause 5.3 for the definition of a CSB UE;
- IMS networks and IMS UEs without CSB support should not to be impacted;
- CS core, PS core, GERAN, UTRAN (incl. TS 24.008) are not to be impacted. Conclusively, changes should be restricted to the IMS elements and the UEs that support CSB for IMS.
- Protocols connecting the IMS to the CS domain, to the PSTN and to other SIP networks, including other IMS networks should remain unchanged.
- CS only UEs and PS only UEs are not to be impacted;
- The use of CS bearers in the context of IMS should be transparent for the user.
- CSB UE provides capabilities to bind the corresponding CS and IMS sessions for the user. Note that some network capability might also be required to achieve this.
- Regardless of the bearer used for a session (CS bearer or voice over IP bearer), the same service logic and end-user service experience shall apply.
- In case of voice services, the overall service quality as perceived by the users shall be the same or better than the one of R99 CS services. This includes that set up times of a speech call shall not be longer than those experienced e.g. in a R99 CS call.
- The coverage and voice quality provided by CS bearers in the context of IMS is assumed to be the same as the coverage and voice quality provided by GERAN (and/or UTRAN) CS voice coverage. Moreover, in case of handover into non-DTM or non 3G areas, voice service shall be preserved.
- The quality of the voice call (e.g. voice quality, setup delay, handover, etc…) shall not be impacted from a user perception point of view regardless of whether the CS call is combined with an IMS session or not.
- The use of CS bearers in association with an IMS session for a UE requires that the UE is CS attached and IMS registered.
- There are no additional authentication mechanisms required.

5.2 Architectural Requirements

- It shall be possible to perform correlation of charging that is performed in the CS, PS and or IMS nodes within the context of using CS bearers for IMS (both online and offline). This shall ensure consistent end-user charging.
- The architectural solution supports interworking with conversational IMS services, which use PS bearers (see also subclauses 5.3 and 5.4);
- The architectural solution supports migration towards conversational IMS services, which use PS bearers;
- The architectural solution supports handover scenarios, including inter-system handover;
- The architectural solution supports roaming scenarios with home GGSN ("IMS with GPRS roaming");
- The architectural solution supports roaming scenarios with visited GGSN ("IMS roaming");
- The architectural solution does not unnecessarily prohibit end-to-end compressed speech;
- The architectural solution is compatible with the IMS home control paradigm;
- According to the addressing principles as defined in subclause 7.5.1 of TS 22.228 [1] it shall be possible for the network operator to use the same E.164 number for IP multimedia sessions and CS speech telephony or different E.164 numbers for CS speech telephony and IP multimedia sessions.

5.3 Interworking between different terminals

When analyzing the different session scenarios and migration aspects, the following terminal types should be considered (strictly within the context of this TR) from interworking point of view:

- IMS VoIP capable UE: an IMS terminal that supports both VoIP bearers and CS bearers for SIP/IMS voice, but prefers IMS VoIP for voice when originating sessions.
- CSB UE: a terminal that is capable to use a CS bearer for the voice component of an IMS session. It can be either a UE capable of supporting services in a CS/CSB approach (CS/CSB UE) or a UE capable of supporting services in a IMS/CSB approach (IMS/CSB UE)
- SIP/IMS VoIP only UE: a terminal that is capable of supporting SIP/IMS voice using VoIP bearers only.
- CS Domain/PSTN endpoints.

5.4 Session scenarios

To guide the design of solutions and determine their feasibility the following scenarios shall be considered:

Calls between CSB capable UEs:

- CSB UE(A) initiates a voice session to CSB UE(B) using IMS signalling;
- CSB UE(A) initiates a multi-component (voice + MSRP) session to CSB UE(B);
- CSB UE(A) initiates a voice session to CSB UE(B), then adds MSRP component;
- CSB UE(A) initiates an MSRP session to CSB UE (B), then adds voice component;

Calls from a CSB UE to a SIP/IMS VoIP UE:

- CSB UE(A) initiates a voice session to IMS VoIP capable UE(B) using IMS signalling;
- CSB UE(A) initiates a multi-component (voice + NRT) session to IMS VoIP capable UE(B);
- CSB UE(A) initiates a voice session to IMS VoIP capable UE(B), then adds MSRP component;
- CSB UE(A) initiates an MSRP session to IMS VoIP capable UE (B), then adds voice component;

- CSB UE(A) initiates a multi-component (voice + NRT) session to SIP/IMS VoIP only UE(B);
- CSB UE(A) initiates a voice session to SIP/IMS VoIP only UE(B), then adds MSRP component;
- CSB UE(A) initiates an MSRP session to SIP/IMS VoIP only UE(B), then adds voice component;

Calls from a CSB UE to CS Domain/ PSTN endpoints:

- CSB UE(A) initiates a voice session to a CS Domain/PSTN endpoint;

Calls from a SIP/IMS VoIP UE to a CSB UE:

- SIP/IMS VoIP only UE (A) initiated a voice session to CSB UE(B)
- IMS VoIP capable UE(A) initiates a multi-component (voice + MSRP) session to CSB UE(B);
- IMS VoIP capable UE(A) initiates a voice session to CSB UE(B), then adds MSRP component;
- IMS VoIP capable UE(A) initiates an MSRP session to CSB UE(B), then adds voice component;

Calls from a CS Domain/PSTN endpoint to a CSB UE:

- CS Domain/PSTN endpoint initiates a voice session to an CSB UE(B);

Session scenarios between CSB UEs and terminals that lack capability for CSI shall be considered as well.

5.5 Solution characterisation

Solutions proposed for consideration in this TR are characterised in part by the high level objectives they aim to meet. In particular, for each candidate solution, the following questions must be answered:

- How is charging correlation between CS domain and IMS provided, if it is provided at all
- How is control of CS bearers with IMS SIP signalling provided, if it is provided at all
- What capability exchange mechanisms, if any, are proposed (with or beyond the standardised presence service).
- How is it determined whether and when to invoke the combination of IMS and CS bearers. For example, will it be a capability the UE decides to use depending on application request, or the network based on its capabilities?
- Is the proposal a network option, a terminal implementation option or an option at some other level ?
- To which extent does it require standardisation beyond existing standards?

6 Architecture alternatives

Editor's Note: This section will describe the considered alternatives

6.1 Alternative A

6.1.1 Architecture principles

This alternative supports two modes for establishment of the Circuit-switched bearer:

- End-to-end, in which the bearer is established between two CSB UEs
- End-to-Gateway, in which the bearer is established between a CSB UE and a Media Gateway

Significantly, the procedures for these two modes are the same at both UEs – the mode is determined by UE and network capabilities.

In the second case, then as far as the peer IMS client is concerned, the session may be a standard IMS session.

Some further principles are:

- For end-to-gateway Circuit Switched Bearers, the call flows follow IMS Release 5. Where R5 IMS call flows include a PDP Context set-up, the CSB flows include a circuit switched bearer setup
- The Circuit Switched bearer is considered to be a single media component within the SIP session. The use of CSB may be indicated in the Session Description (SDP). The indication must include enough information to establish and identify the Circuit Switched bearer associated with the session, both to the UEs and to other systems (e.g. billing systems). It is assumed this can be done using the Called and Calling Party Numbers.
- No impacts to CSCF functionality – network control is provided by an Application Server; in this case initial filter criteria – possibly taking into account the media description - need to be in place to route the session to the AS.
- No impacts to MGCF or Media Gateway functionality

6.1.2 Architectural components

Three architectural components are defined to support Circuit Switched Bearers. The components will be located at existing IMS elements, specifically the UEs and Application Servers. The location of each of the components is negotiated at session establishment. The components are:

- "Circuit Bearer Control Function (CBCF)" – this performs a variant of third party call control [3] over the establishment of a Circuit Bearer for use at one end of a SIP session
- "Circuit Bearer Originating Function (CBOF)" – originates the Circuit Bearer
- "Circuit Bearer Terminating Function (CBTF)"– terminates the Circuit Bearer

Note that the Circuit Bearer Originating and Terminating functions may be either SIP UAs or the CS domain portion of the UE. In the case that the CB(O/T)F is a SIP UA then it routes a session through a standard MGCF/MGW in order to establish the Circuit Bearer portion of the session.

These functions are illustrated in the figure below:

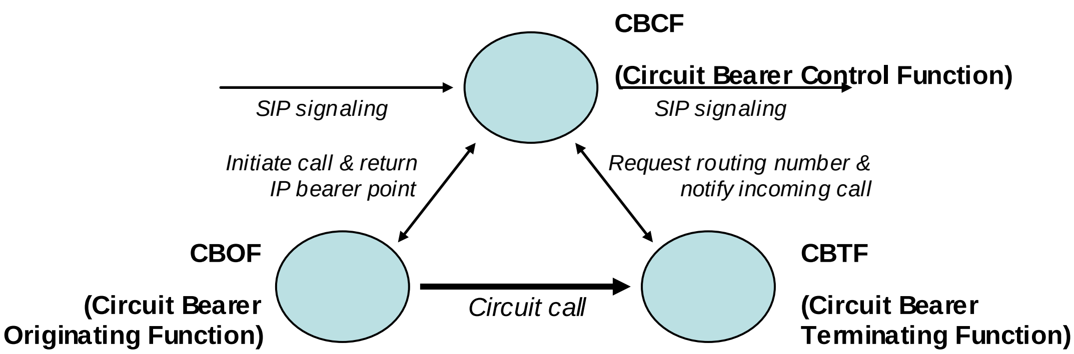

Figure 6.1.2-1: Architectural components for Alternative A

The operation of these three functions is described in more detail below:

The **Circuit Bearer Control Function**:

- Determines that a Circuit Bearer is required for a media component of a SIP session
- Obtains a routing number (E.164 number) for the Circuit Bearer from the Circuit Bearer Terminating Function and provides this to the Circuit Bearer Originating Function
- Obtains a source number (E.164 number) for the Circuit Bearer from the Circuit Bearer Originating Function and provides this to the Circuit Bearer Terminating Function
- Requests the Circuit Bearer Originating Function to originate a Circuit Bearer

- Received a notification from the Circuit Bearer Terminating Function that the Circuit Bearer is established
- In the case that the CBOF or CBTF is a SIP UA function within the network, i.e. in the end-to gateway mode, the CBCF obtains the media parameters for the Circuit Bearer (i.e. IP address and port on the Media Gateway) and provides these to the peer SIP UA within the Session Description. Thus, in the end-to-gateway mode, the CBCF applies mechanisms similar to third party call control [3] to establish the bearer between the media gateway and the peer SIP UA. This principle is illustrated in figure 6.1.2-2.

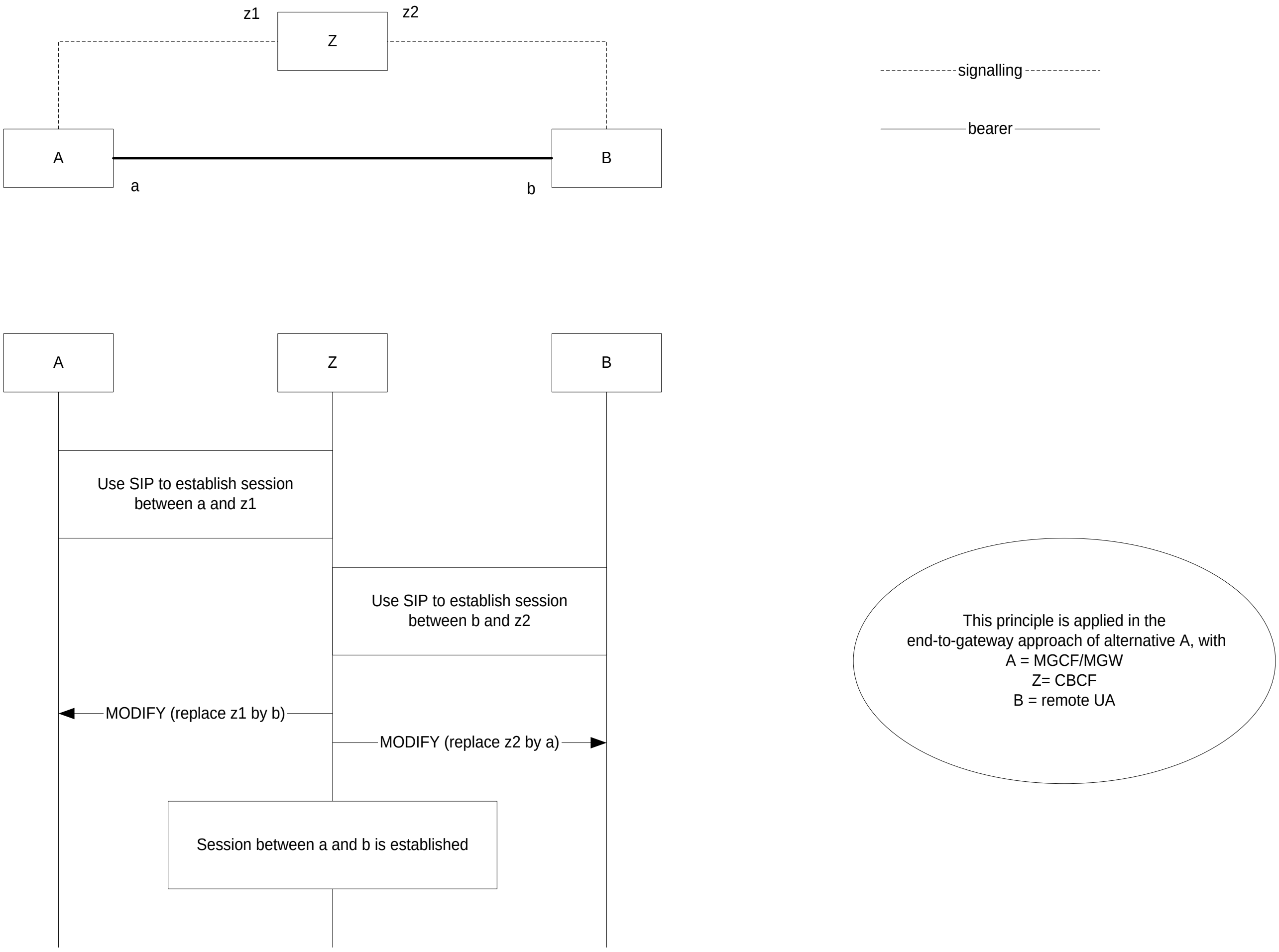

Figure 6.1.2-2: Using a 3pcc-like mechanism from the CBCF

The **Circuit Bearer Originating Function**:

- Receives a request from the CBCF to originate a Circuit Bearer to a provided E.164 routing number
- Provides a source number (E.164 number) to the CBCF
- Originates the call
- In the case that it is a SIP UA within the network, provides the media parameters (i.e. IP address and port) for the Circuit Bearer to the CBCF

The **Circuit Bearer Terminating Function**:

- Receives a request from the CBCF to terminate a Circuit Bearer from a provided source number (E.164 number)
- Provides an E.164 routing number to the CBCF
- Receives the call
- In the case that it is a SIP UA within the network, provides the media parameters (i.e. IP address and port) for the Circuit Bearer to the CBCF

The means of communication between the CB(O/T)F and the CBCF depends on the location of these functions. If they are co-located, then this is internal communication. If they are not co-located then the information is carried within SIP/SDP.

6.1.3 Architecture configurations

6.1.3.1 Configurations Overview

The following configurations are supported:

1) End-to-Gateway

In this case, the entire mechanism is local to one user's end of the session. The mechanism could be duplicated at the peer UE, or this could be a standard VoIP UE.

The Circuit Bearer Control Function may either be within the network (Application Server) or within the UE. This gives rise to "network-control" and "client control" models.

The Circuit Bearer Originating and Terminating functions are either both at the client (client control model) or at the client and Application Server (network control model). The Circuit Bearer can be established in either the client-to-network direction or network-to-client direction. This choice is independent of the session direction. See subclause 6.1.11 for more details.

In these configurations, the circuit bearer is established between the UE and some local gateway at the same point in the call flow as a PDP Context would usually be established. The Circuit Bearer Control Function acts as a third party call controller and a back-to-back user agent in order to mediate between the SIP call legs between the CBCF and UE, gateway and peer client.

In the IMS domain, in the session to get CS call SDP, the destination E.164 number in Tel: URL will always point to the CS part of CSI UE. It may need some indication in the SIP message that the call will route to the CS domain.

In the CS domain, the destination E.164 number can be configured as follows:

- In the network control mode, the destination E.164 number will point to the CBTF. It is used as a PSI in the IMS domain. Different users can share the same destination number, which can reduce the configuration work in the CS domain and the IMS domain.
- In the client control mode, the destination E.164 number will always point to the IMS part of CSI UE. Different users cannot share the same destination number. The destination number needs to indicate that the call must change domains. The destination E.164 number may use a different number than the user had in the CS domain. It can use a Prefix-number or some other method.

2) End-to-end

In this case, all capabilities are provided by the clients. The Circuit Bearer Originating and Terminating functions are provided by the two UEs.

In the CS domain, the destination E.164 number will always point to the callee.

6.1.3.2 The end-to-gateway configuration, network control model

The following figure illustrates the end-to-gateway configuration with network control:

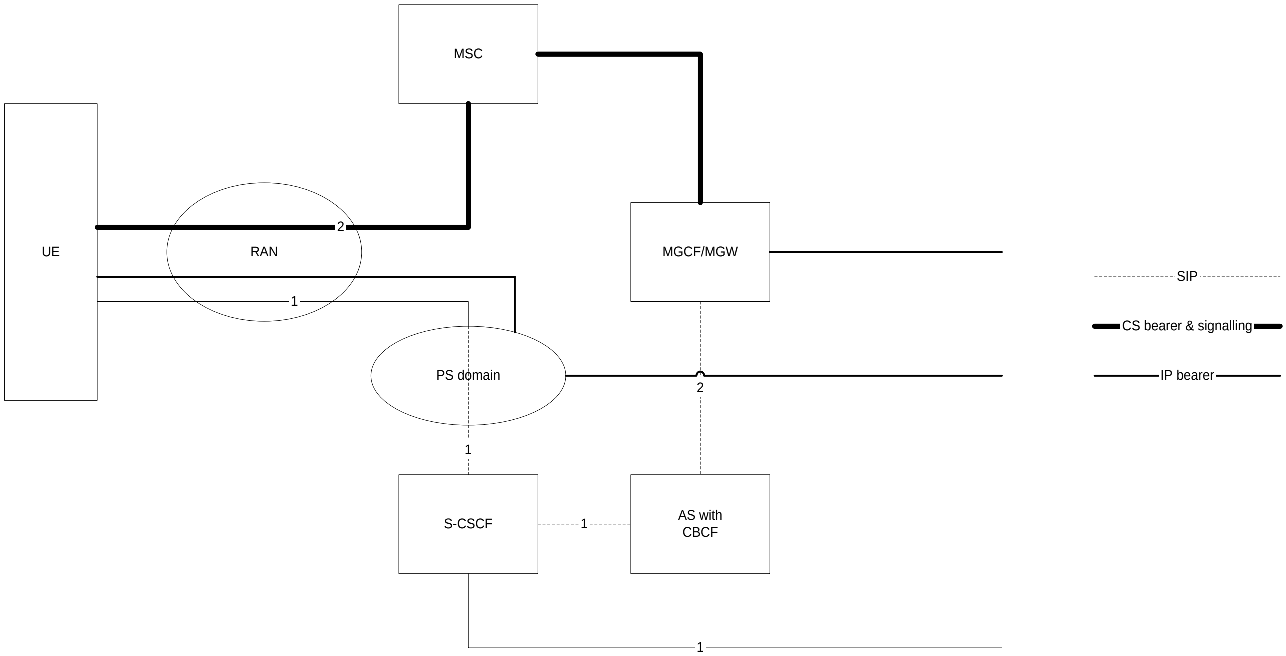

Figure 6.1.3.2-1: End-to gateway, network control

In this approach, the CBCF is located in an AS. The SIP INVITE towards the remote UA (1) is routed through the Application Server with CBCF. Then the CS bearer is established (2) making use of CS/IMS interworking in the MGCF/MGW (2). If the CS call is established in network-to-client direction, then routing tables have to be configured correctly to ensure that the call reaches the CS part of the UE (and not the IMS part). The variant of 3pcc illustrated in figure 6.1.2-2 ensures that the IP traffic is routed efficiently.

The solution requires circuits (resources) in both the MSC and in the MGW. On the call control layer, MSC, MGCF, S-CSCF and CBCF keep state.

6.1.3.3 The end-to-gateway configuration, client control model

The following figure illustrates the end-to-gateway configuration with client control:

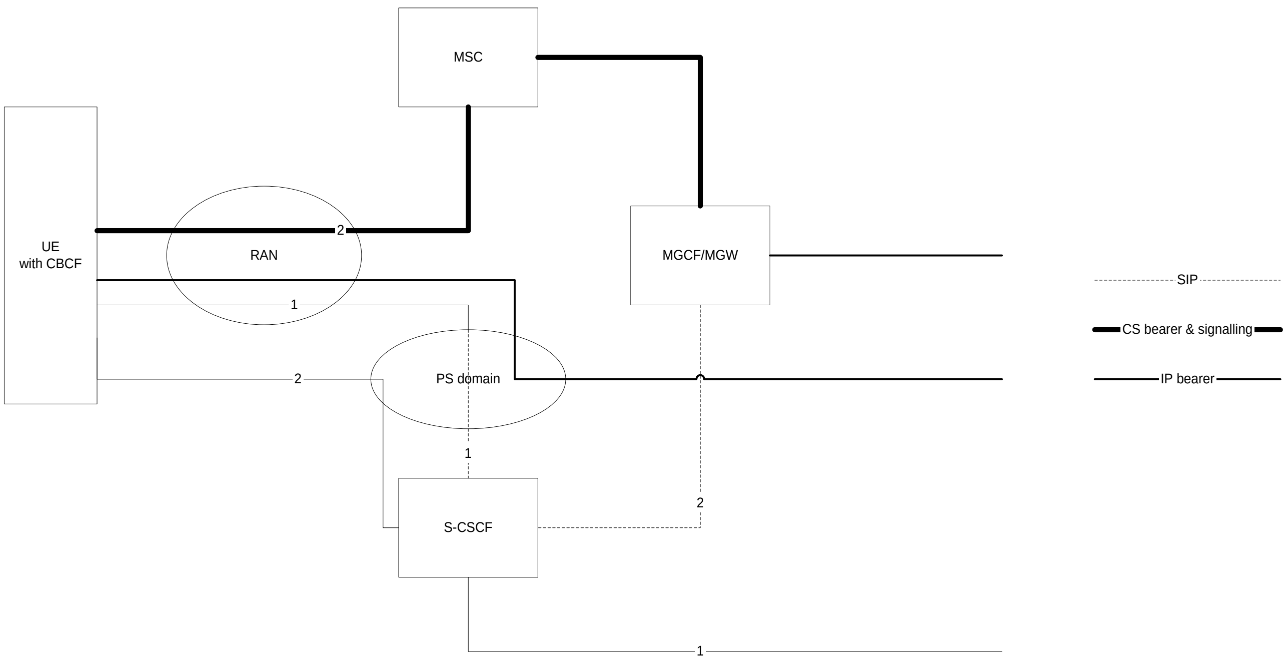

Figure 6.1.3.3-1: End-to gateway, client control

In this approach, the CBCF is located in the UE. The UE sends a SIP INVITE towards the remote UA (1). Then the CS bearer is established making use of CS/IMS interworking in the MGCF/MGW by establishing a call from the UE to itself in a loop through CS domain and IMS (2). Routing tables have to be configured correctly to ensure that the call reaches the "right part" of the UE. The variant of 3pcc illustrated in figure 6.1.2-2 in subclause 6.1.2 ensures that the IP traffic from the remote party is routed to the MGW.

As in the network control case, the solution requires circuits (resources) in both the MSC and in the MGW. On the call control layer, MSC, MGCF, S-CSCF and CBCF keep state.

In the client control case the loop from the UE through the network back to the UE puts additional burden on the air interface, just to discover an IP address and a port number. Thus this configuration is not recommended for standardisation.

6.1.3.4 The end-to-end configuration

In the end-to-end configuration, the SIP INVITE is sent between the two UEs using the usual IMS mechanisms. Indications in the message trigger the end-to-end establishment of a CS call between the two UEs. This requires that both UEs support the mechanism.

To support charging models such as a combined rate for a CS call with a parallel IMS session, additional mechanisms would be required to provide the required charging correlation. (Note that the indication in SIP as such is not sufficient as it could be forged by the UE.). As of now the network does not have mechanisms to verify that the parallel CS call has indeed been established, that the reduced rate, e.g. for the IMS component, is used only in parallel at the same time and towards the same destination with the CS component.

Under the assumption that such charging models are required, the end-to-end configuration requires additional network capabilities before it can be recommended for standardisation.

The end-to-end configuration is described in more detail and illustrated in subclause 6.1.6.2 below.

6.1.4 Negotiation

The location of the various functions can be negotiated as session setup. Furthermore, if the UE does not provide the CBCF itself, it needs not be aware of whether the Circuit Bearer Control Function is provided by the network or by the peer client.

Negotiation is based on a new capability to indicate Circuit Bearers within the SIP messages. A possible place is the Media Component of a Session Description. This indication includes an E.164 address (possibly in the ‘c=’ line) and an indication of whether the sender wished to originate or terminate a Circuit Bearer (possibly using the “comedia” draft).

Note: While the SDP is a possible place for the indication within SIP signalling, this is a detail, which does not preclude a different solution by stage 3 protocol design, if alternative A is standardised. However, for simplicity, the SDP based solution is assumed in the remainder of the subclause.

A UE which is capable of supporting the Circuit Bearer Originating or Terminating Function indicates its support in an SDP offer. If the UE is also capable of supporting VoIP, it may offer this as well.

Furthermore, if the UE supports the CBCF itself, then it is capable of making a Circuit Bearer look to the peer like a VoIP session (End-to-Gateway mode with client control). In this case it may also offer a VoIP session.

The Circuit Bearer Control Function recognises the offer to use a Circuit Bearer. It will contact a Circuit Bearer Terminating or Originating Function (as appropriate) and obtain the required VoIP details. These are used to replace the CSB offer in the SDP before forwarding to the peer client.

On receipt of the answer from the peer client, the CBCF removes the VoIP SDP and passes this to the local CB(O/T)F. It replaces the VoIP SDP with a CSB reply before sending the answer back to the originating client. The CB(O/T)Fs are then in a position to establish the Circuit Bearer.

The originating client recognises that a CBCF is available by the presence of a valid CSB answer in the SDP. The originating client need not be aware of whether this CBCF is provided by the network (End-to-Gateway mode) or by the terminating client (End-to-End mode). Obviously, it will only be possible for the CBCF at the terminating client to be triggered if the network has allowed the CSB SDP to pass unchanged from end-to-end. Thus the network always has the option to trigger its own CBCF if it so wishes.

The negotiation between originating client, CBCF/CBTF, MGCF/MGW and terminating client is shown below. Remember that the CBCF/CBTF/MGCF/MGW may be in the terminating client, in which case the interactions between CBCF and terminating client are internal – there may be no actual VoIP parameters. However, from the originating client point of view, the interactions are the same.

Conversely, the CBCF/CBTF may be in the originating client, in which case the terminating client and MGCF/MGW do not see any difference in the interactions.

In this way a number of different deployment modes can be supported without introducing options into the protocols.

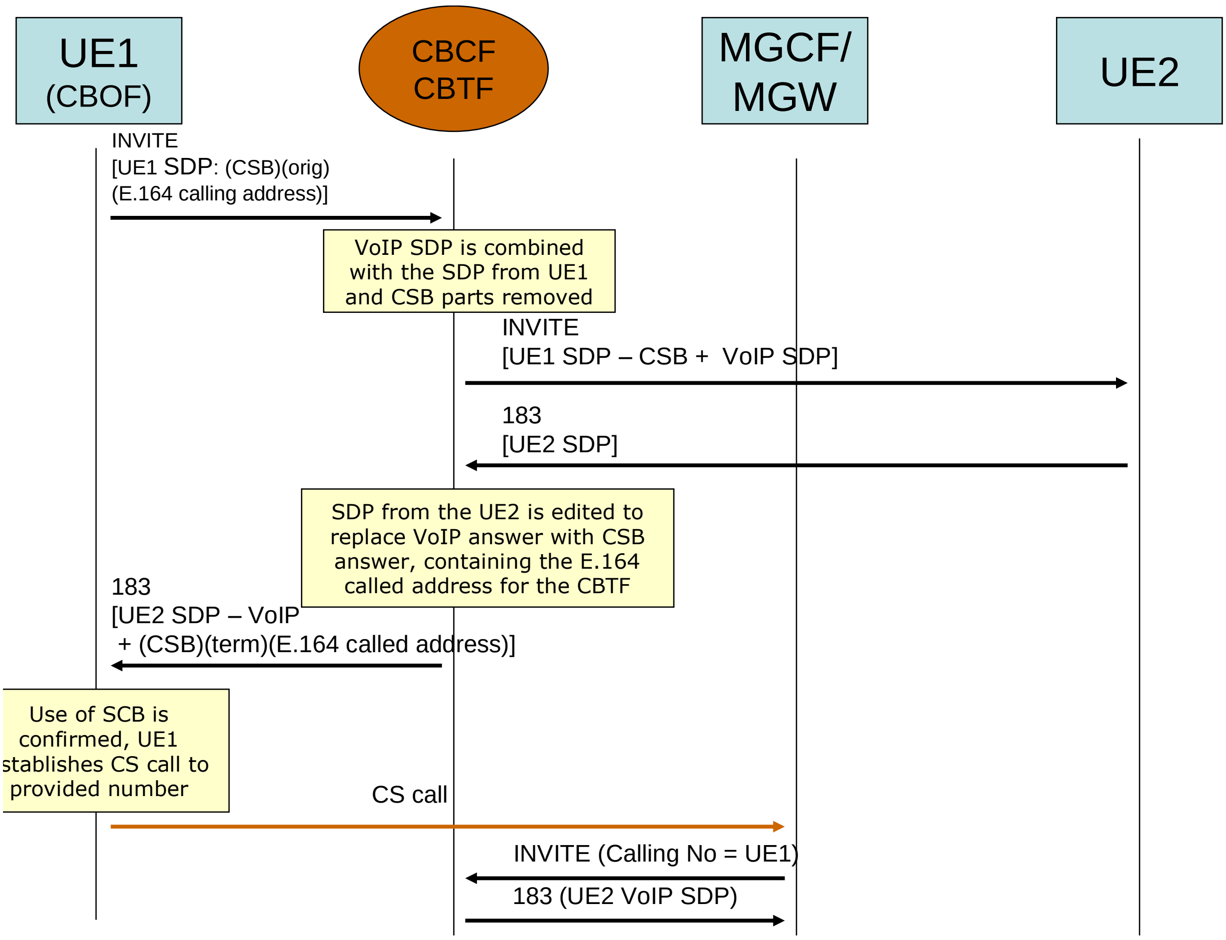

Figure 6.1.4-1: High level flow for Alternative A

6.1.5 Use of preconditions

SIP Pre-conditions can be used in association with Circuit Bearers in just the same way as with PDP Contexts. The SDP associated with a Circuit Switched bearer will be marked as 'pre-conditions not met' until the Circuit Switched Bearer is established and answered.

Note that for simplicity, we assume that Circuit Switched Bearers in the End-to-Gateway case are Answered immediately. It is assumed that charging for the session will be based on correlation of the IMS session with the Circuit Bearer(s) and thus the true time of Answer will be known. With some additional complexity, it would be possible to align the CSB answer with the SIP answer, although there may then be speech clipping issues.

6.1.6 Correlation of charging for the Circuit Switched Bearer

6.1.6.1 End-to-end case

Charging information in both the CS and IMS domains will include identities of the involved parties. This information, along with timing information, could theoretically be used to identify the CS call and IMS session as 'combinational' and apply the appropriate rating rules.

Called and calling party number information included within the SDP may be used at least to determine that associated CS charging information should exist. Note that this information cannot be trusted as a definitive indication of combinational services – since it could be forged by the UE – however, it can be used to make correlation more efficient since the charging systems need not search for associated CS domain charging information when these indications are not present.

However, correlation on this scale is probably not practical.

6.1.6.2 End-to-gateway case

All end-to-gateway client-to-network CS calls are directed to a special number allocated to the CBTF. These can therefore be readily identified within the charging systems.

End-to-gateway network-to-client CS calls are originated from a special number allocated to the CBOF. These can therefore be readily identified within the charging systems.

A simple approach would be to zero rate these calls and charge at the IMS layer (noting that there is no fraud opportunity since unexpected calls to the CBTF can be rejected immediately and management action taken).

If the user is roaming (and there is no IMS roaming in place), then the end-to-gateway call will be routed from visited to home network and charged by the visited network. The IMS charging systems will need to detect that the user is roaming and apply the appropriate charge at this layer. However, the inter-operator charges levied by the visited network will be according to the CSB call setup direction (client-to-network vs network-to-client) not according to the IMS setup direction.

6.1.7 Use of pre-existing CS domain bearers

6.1.7.1 General

CSB UEs may indicate that they have a pre-existing CS bearer which they believe to be suitable for a real-time component of the session. This can occur in two cases, corresponding to the end-to-end and end-to-gateway cases discussed above.

6.1.7.2 End-to-end case

This corresponds to the case in which the CSB UE has a normal CS domain call (in any state) and wishes to associate an IMS session with this call. The UE may indicate in the CSB SDP that it has such a call.

The terminating UE must be able to receive such indications and associate the existing CS domain call with the session. Subsequently, the UEs behave just as if the CS domain call had been established as part of the IMS session establishment.

It should also be possible for the originating UE to establish both IMS session and CS call in parallel. In this case, the same indication is provided to indicate in the SDP that a call is already present. At the terminating UE, however, the IMS session may arrive first, in which case the UE needs to recognise the indication as a request to wait for the CS call about to arrive (in exactly the same wayas the end-to-end case described above).

6.1.7.3 End-to-gateway case

This corresponds to the case in which the CSB UE has a CS domain call associated to an IMS session and that session is to be replaced by a new one (for example due to a SIP REFER or an incoming INVITE containing an Replaces header).

The CSB UE may indicate in the SDP that is has an existing CS domain call which it wishes to associate it with a new SIP session. The CBCF determines whether the existing call is suitable for association with the new session and returns SDP to the UE indicating either that the association was successful, or that a new call is required.

6.1.8 SDP extensions

SDP extensions required for this mechanism may be defined by 3GPP as new SDP attributes without reference to IETF.

Alternatively, if new Network and Address types are preferred, an Informational RFC is required.

This decision is a Stage 3 issue.

6.1.9 Interaction with CS Supplemetary Services

6.1.9.1 End-to-gateway case

Interaction with outgoing call barring supplementary services and outgoing CAMEL services needs to be considered. However since the call is to a special routing number, it should be possible to bypass these.

6.1.9.2 End-to-end case

We can assume in this case that any CS domain supplementary services operate in the usual way. This makes it important to ensure that any active CS domain services have equivalent or similar IMS services. For example, if the user has CS Call Forwarding active but no similar service in the IMS domain, then the CS call and the associated IMS session may terminate at different endpoints.

In the case of pre-established end-to-end CS calls, this limits the call routing services to those available in the CS domain.

In the case that end-to-end CS calls are established after initial IMS session negotiation, IMS service logic can be used to route the call and (subject to privacy restrictions) the correct routing number for the associated CS call provided.

6.1.10 Routing considerations

- For the case of IMS sessions to be associated with a pre-existing CS domain call, both CS calls and IMS sessions must be 'diallable' using the same E.164 number – that means the same E.164 number must route within both CS domain and IMS domain, and the call must stay within the domain it starts in.
- For the case of end-to-end CS calls made as a result of negotiation within an IMS session, the same approach in which calls stay within the same domain is required.
- For the end-to-gateway, client-to-network case, each operator allocates a single number or number range for routing towards the CBTF.
- For the end-to-gateway network-to-client case, then some mechanism is needed to route from the IMS to a given user in the CS domain.

6.1.11 Roaming Considerations

6.1.11.1 End-to-gateway configuration

Assume the UE is roaming in a visited network different from the home network.

If the CS bearer is established in the client-to-network direction, the destination of the call establishing the CS bearer is determined by a phone number in the home network; following CS routing principles, the call will be established through the home network. As a result the CS bearer will be routed through the home network in all cases, i.e. also on the originating side. This is a disadvantage. Alternatively, routing through a gateway in the visited network could be achieved by appropriate agreements between home and visited networks on the routing of certain E.164 numbers directly to the IMS, rather than through the international PSTN. However, this implies an additional operational overhead.

If the CS bearer is established in the network-to-client direction, then the S-CSCF in the home network will pass the call to a BGCF. The BGCF could possibly select an MGCF in the visited network. This would avoid the need to route the call through the home network. Note that the support of visited-network break-out implies a second entry point to the IMS in the visited network and thus adds operational effort in terms of configuration and security.

6.1.11.2 End-to-end configuration

In the end-to end configuration the CS bearer is routed as a usual CS call, i.e. usually routed through the home network on the terminating side, but not routed through the home network on the originating side.

6.1.12 Direction of CS bearer establishment in the end-to gateway configuration

In the end-to-gateway configuration the CS bearer could be established in both the network-to-client and client-to-network direction. In the network control model, network-to-client means that the AS hosting the CBCF calls the (CS part of the) UE, while client-to-network means the (CS part of the) UE calls the AS. In the client control model, network-to-client means that the IMS part of the UE calls the CS part of the UE, while client-to-network means the CS part of the UE calls the IMS part of the UE.

For the network-to-client direction:

- it is probably desirable to have special handling of the incoming call in the UE, e.g. do not ring for a CS bearer, which is established as part of an originating SIP session,
- the call is established as a terminating CS call. Thus it has to be ensured that terminating services, like unconditional call forwarding, do not disrupt the CSB logic, but consider alignment of CS and IMS call forwarding,
- if necessary, incoming call barring might be bypassed. In this case incoming call barring services need to be configured to allow calls from the CBOF in the network,
- the possible need to page the terminal, and for the terminal to establish a signalling connection with the CS domain in order to receive the incoming call, can take some time which will contribute to the overall call setup delay,
- flexible charging models are probably easier to realise with the network-to client direction. The network can easily "zero-rate" the CS bearer in the network control case, also in case of online charging.

For the client-to-network direction:

- call flows are more aligned to the PS domain handling, but note that there is a lack of Go or Gx interface, instead network control of the bearer requires that the AS can reject unauthorised call attempts,
- the Application Server provides the UE with a number allocated to the CBTF. Routing tables in the CS domain are configured to route the CBTF number to the IMS, and the IMS is configured to route it towards the Application Server. This requires definition, configuration and administration of a range of specific phone numbers for that purpose; it needs to be ensured that these will also work in roaming scenarios,
- for roaming scenarios, there is an issue with routing the bearer, which may impact call-set-up time, see subclause 6.1.11,
- to avoid abuse of the specific number range, the CBCF needs to verify that the CBTF number is indeed used by the correct UE only,
- as above, interactions with services in the CS domain needs to be considered. The main problematic service could be call blocking/closed user group services. The phone number chosen for the CBTF needs to be such that it will be allowed by these services. CAMEL originating side services need to be aware of the CBTF number and ensure calls to this number are allowed to proceed as desired,

- routing of the call directly from the VMSC to the MGCF/MGW may be possible. This may be more efficient if the network is of a scale with many gateways. By contrast, network-to-client establishment requires the call to be routed through a GMSC.

A UE, which supports the end-to-end configuration, should also be able to support both the network-to-client direction and client-to-network direction.

6.1.13 Example call flows

6.1.13.1 General

This section presents a number of example call flows based on the above mechanism.

In all the call flows, we have assumed that the Circuit Bearer is initiated in the client to network direction for the End-to-Gateway cases. All the flows can be redrawn with Circuit Bearer establishment in the network to client direction.

For simplicity, the IMS and CS domain network elements (CSCFs, separation of MGCF/MGW, BGCF, MSCs etc.) are not shown, since these are not significantly impacted by the mechanism. Call flows including these additional elements are ffs. It is assumed that IMS routing and ISC mechanisms can be enhanced to support routing from the S-CSCF to the AS hosting the CBCF in the network control case.

In particular, leaving out the S-CSCF allows to take the same call flows apply for the "network control" (CBCF in an Application Server) and "client control" (CBCF in the terminal): Note that in the client control case, some of the procedures are implemented within the client, and so are not seen externally.

6.1.13.2 Originating user with Circuit Bearer, End-to-Gateway

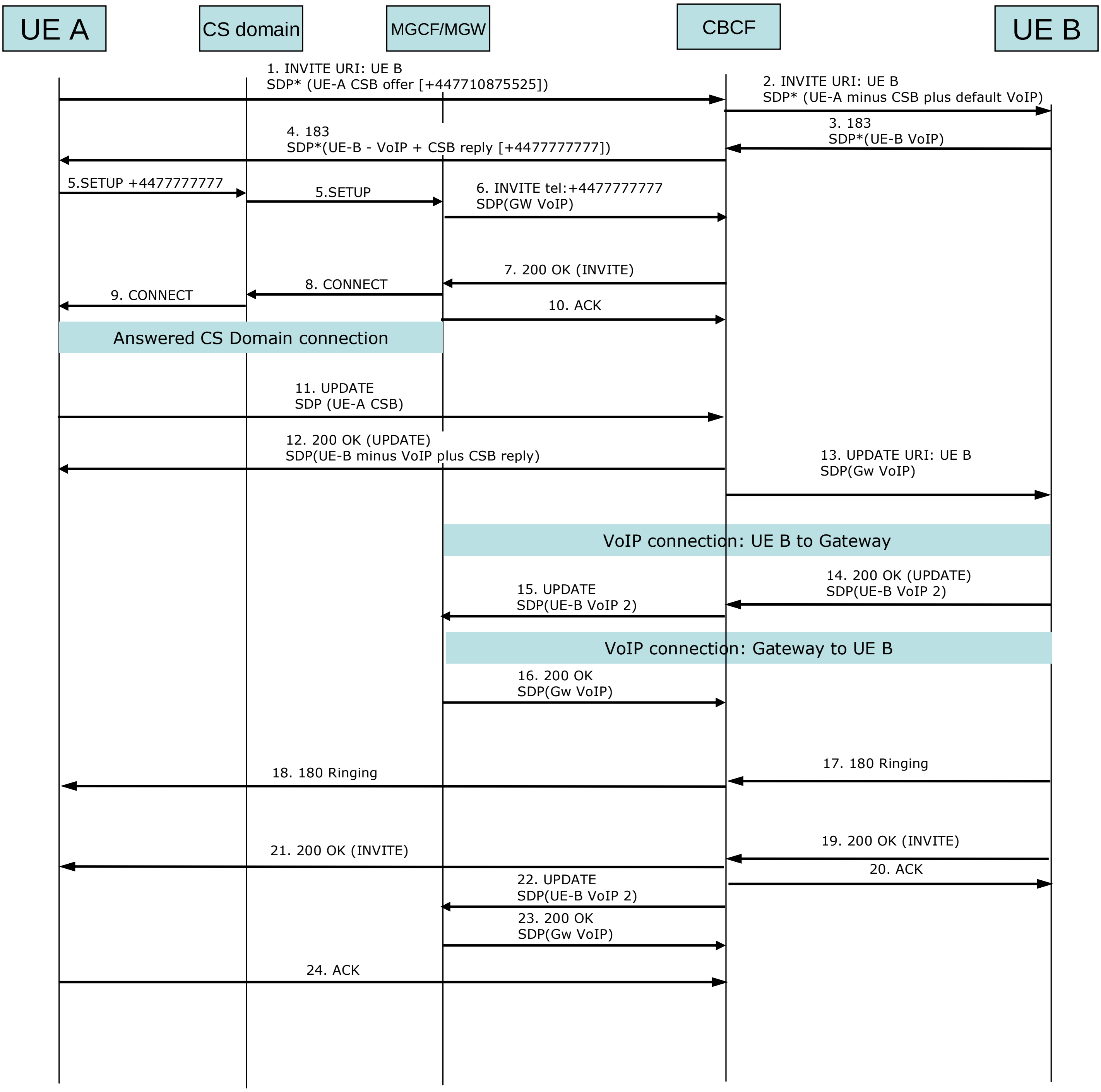

Figure 6.1.13.2-1: Call Flow for originating terminal end-to-gateway case

6.1.12.3 Terminating user with Circuit Bearer, End-to-Gateway

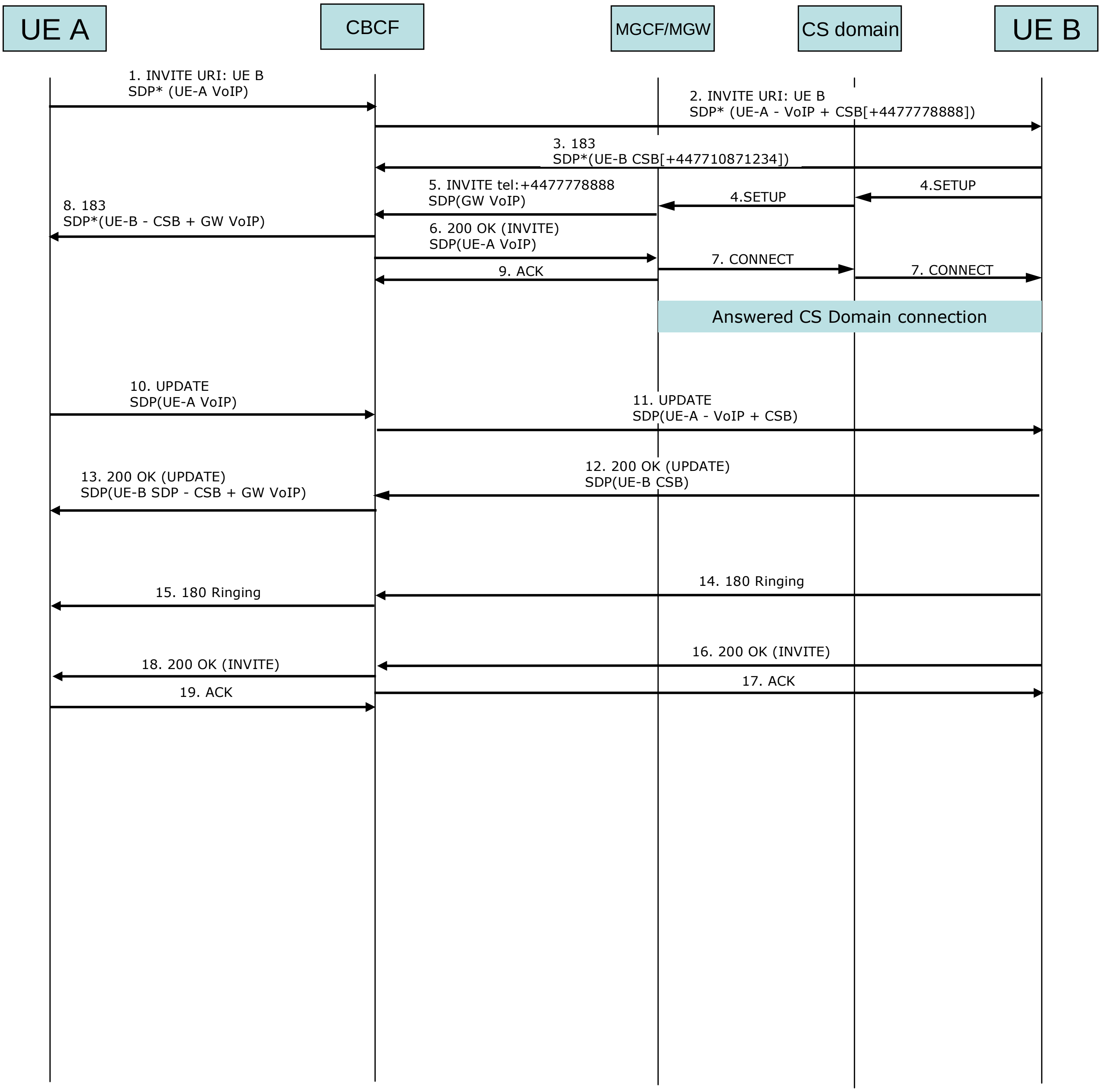

Figure 6.1.13.3-1: Call flow for terminating user end-to-gateway case

6.1.13.4 End-to-end case

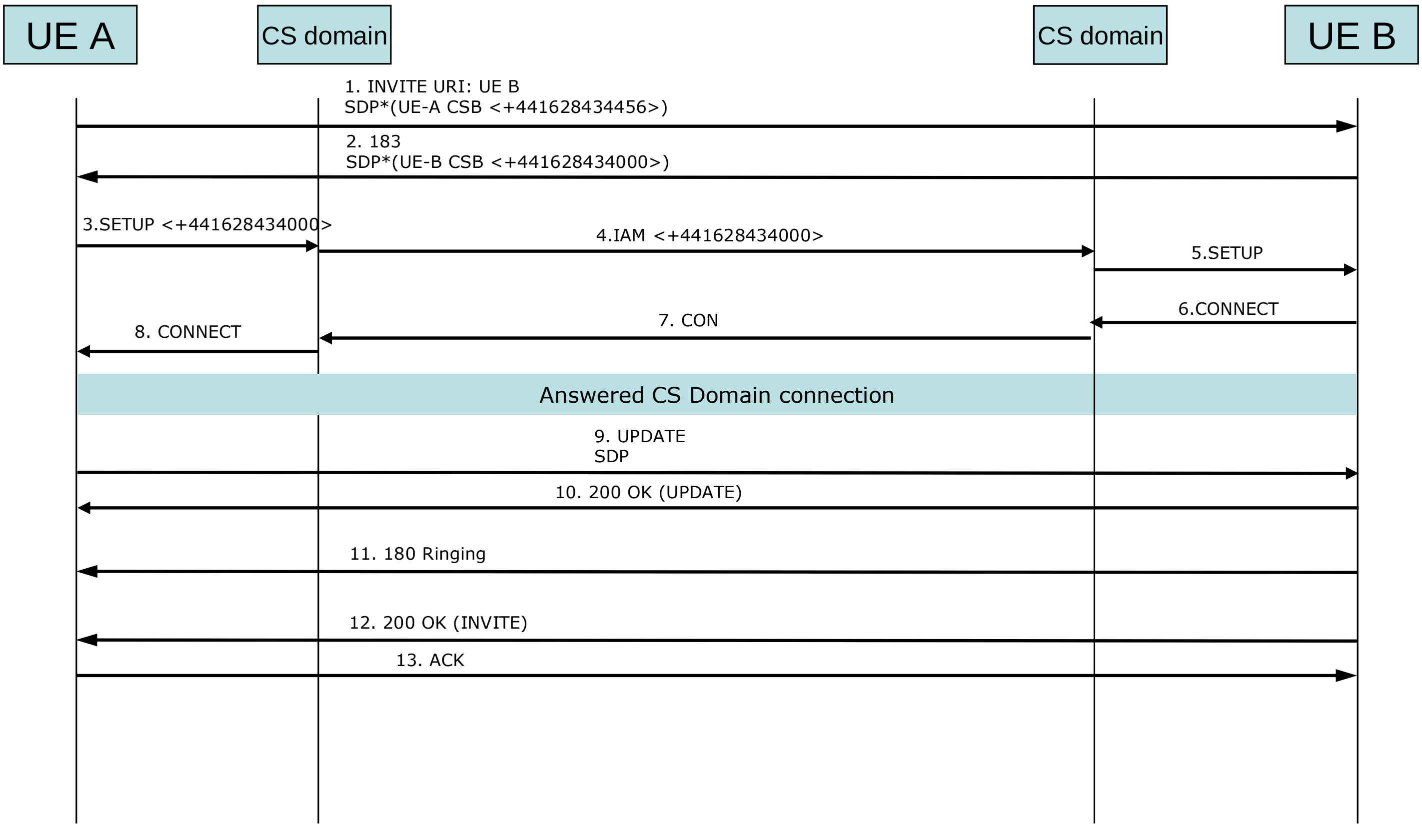

Figure 6.1.13.4-1: Call flow for end-to-end case

It should be noted that the procedures at UEs A are the same as for Section 6.1.6.2. If the End-to-Gateway flows with network-to-user establishment of the circuit-switched call are also supported, then the procedures here for UE B would be the same as for a terminating UE supporting those flows.

The direction of establishment can be negotiated by an indication in the SDP (the comedia draft would be suitable for this). In this way, a single set of procedures at the UEs can handle both the End-to-end and End-to-Gateway cases.

6.1.14 Summary of Alternative A

This section presents an architectural option for use of CS bearers with IMS with the following properties:

- A CS call may be associated with an IMS session to provide a real-time bearer. The CS call may be
- established under control of a network-based Circuit Bearer Control Function as part of IMS session setup,
- negotiated directly between two end-users (if permitted by the network),
- a pre-existing CS call established in association with a previous IMS session
- a pre-existing CS domain call established between two endpoints,

- In the first two cases, sessions are controlled entirely using IMS service logic – end-user service experience should not be affected. In particular, all other IMS capabilities - presence, instant messaging, application sharing etc. – will operate exactly as expected
- No impact on CSCFs, MGCF, MGW functionality

- Either the network, or the client, may control the establishment and use of a CS bearer – supporting early testing/deployment of client-based solutions and later migration to network control
- The CS bearer may be local to the user – that is, the media is interworked to VoIP as quickly as possible – or may be end-to-end between clients
- Use of end-to-end versus end-to-gateway CS bearers (with network control) is transparent to the UE
- The configuration and CS call setup direction are negotiated per session, supporting flexibility in terms of deployment models and evolution

Some further issues remain to be investigated:

- Whether the whole solution or only certain options (e.g. network control, client control, end-to-end, end-to-gateway, client-to-network, network-to-client, …) should be considered for further consideration.

6.2 Alternative B

6.2.1 Architecture Principles

This alternative supports the addition of IMS sessions to an ongoing CS call or the addition of a CS call to an ongoing IMS session. The architecture addresses a number of network features that are required for this solution in accordance of the architecture requirements as described by section 5.1 and section 5.2 in TR 23.899.

The proposed solution addresses the following architecture principles:

- The CS and PS domain are not impacted. This means that the existing CS and PS core network are not changed, provided that the PS core network supports the IMS requirements
- The IMS subsystem is not impacted. In this context the CS call control and the IMS session control are clearly separated.
- The UE CSI client software correlates the IMS sessions and the CS calls at a bearer level within the UE.
- The UE CSI client software, CS services and the IMS services may interact to provide a combinational CSI service.
- The UE CSI client software and the network charging mechanisms may correlate the CS and the IMS charging.
- The UE CSI client software will provide a terminal capability detection mechanism.

Editor's note: The Service Interaction and charging correlation remains for FFS

6.2.2 High-Level Architecture

The high-level architecture consists of the following parts:

- **UE**
  The UE (i.e. CS/CBS UE) needs to support simultaneous CS and PS domain access i.e. DTM and/or multiRAB capabilities and in addition the UE should support combinational service client software e.g. including the capability to present the usage of CS and PS domain within the same context to the user.
- **Radio Access**
  The Radio access Network is not impacted by the Combinational Services. The GSM/EDGE and UTRAN radio access technologies should be supported. It is assumed that DTM and/or multiRAB is supported.
- **PS Core Network (PS)**
  The PS Core network remains unchanged, although it should support the IMS.
- **CS Core Network (CS)**
  The CS Core Network remains unchanged. The CS core network contains MSC/VLR, HLR, and possibly other logical elements according to the 3GPP specifications TS 23.002, 24.008 and 29.002.

- **IM CN Subsystem (IMS)**
  The IMS provides the session control of the combinational services. It also addresses the terminal capability detection mechanism. The IMS core includes Rel-6 capabilities as per TS 23.228. The IMS routes the SIP signalling between the UE (A) and UE (B).

- **Combinational Application Server (CSI AS)**
  The CSI AS handles the control of combinational services, such as charging and authorization. The Combinational Server performs the following CSI related functions:

  - Authorization for accessing the combinational services (e.g. check that the user has a combinational service subscription, if needed);

  - Reporting to the charging system;

  The CSI AS may also perform other IMS related functions such as:

  - Handling service policies for the IMS services, e.g. that the content transferred in the user plane conforms to the Combinational service contract (e.g. maximum image size, image encoding types);

  - Monitoring of IMS Service content transmission.

Editor's Note: The role of the CSI AS in context of charging correlation and interworking with supplementary services/CAMEL services is FFS.

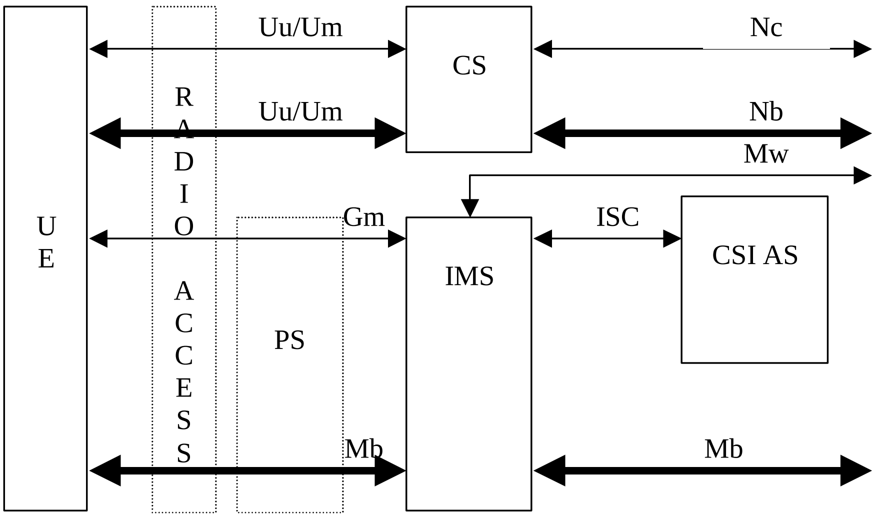

Figure 6.2.2-1: High-Level Architecture of Combinational Services

The high-level architecture consists of the following interfaces as defined in 23.002:

- Interface Gm: UE – IMS Core
- Interface ISC: IMS– Application Server
- Interface Mb: UE – IMS
- Interface Uu/Um: UE - CS Core

In addition to the listed interface, the following inter-operator interfaces are defined in the high-level architecture.

- **Interface Mw: IMS Core operator 1 – IMS Core operator 2**
  The Mw interface supports the signalling plane communication between the IMS Core of an operator and the IMS Core of another operator, for the case where Combinational Service is used between users subscribed to different operators. The protocol for the Mw interface is SIP (as defined by IETF RFC 3261, other relevant IETF RFCs and necessary additions from 3GPP TS 24.228).

- **Interface Mb: IMS Core operator 1 – IMS Core operator 2**
  The Mb interface supports the user plane transport of session mode multimedia content between the IMS of the two operators, in the case where Combinational Service is used between users subscribed to different operators.

- **Interfaces Nb, Nc: CS Core operator 1 – CS Core operator 2**
  The Nc interface supports the communication in the control-plane between the CS Core of an operator and the CS Core of another operator, in the case where combinational services is used between users subscribed to different operators using MAP and ISUP/BICC.
  The Nb interface supports the communication in the user-plane between the CS Core of an operator and the CS Core of another operator, in the case where combinational service is used between users subscribed to different operators.

6.2.3 Functional Architecture

The functional architecture maps the 3GPP functional nodes and interfaces within the high-level architecture. The functional nodes and the interfaces are according to the 3GPP Architecture as described in the 3GPP TS 23.002.

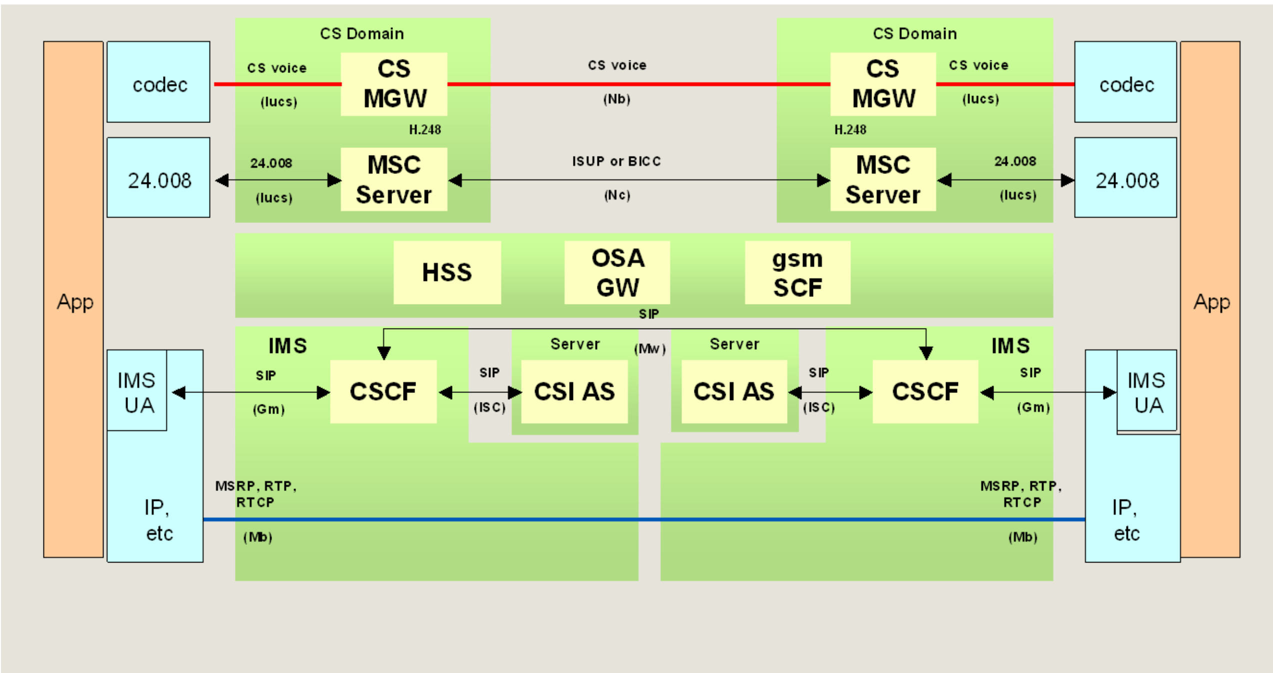

Figure 6.2.3-1: Functional Architecture of Combinational Services

Editor's Note: Any additional interface needed by the CSI AS is FFS.

6.2.4 Correlation of charging for the Circuit Switched Bearer

Charging information in both the CS and IMS domains will include identities of the involved parties. This information, along with timing information, may be used to identify the CS call and IMS session as 'combinational' and apply the appropriate rating rules.

However. Correlation on this scale is probably not practical.

6.3 Alternative C

6.3.1 Architecture principles

The proposed architecture combines IMS sessions with an end to end CS bearer, instead of a PS bearer, to support a voice media.

The basic principle is the capability of the IMS AS in the HPLMN of the originating UE, to perform a 3rd Party Call Control towards both session end points over the CS bearer.

3rd PCC can be invoked by an IMS AS via OSA API or via SIP signalling. An Interworking Function (IWF) in the network is used to map the 3rd PCC into two CS Call Setup (network to client direction). Based on the actual mechanism used for 3rd PCC (OSA API vs SIP signalling), the IWF may be an OSA SCS / MSC or a MGCF. In figure 6.3.1-1, a generic IWF function is used to show how the 3rd PCC is performed, regardless of the actual mechanism used.

The IMS AS in the terminating HPLMN is not involved in the 3rd PCC, but it can still be invoked in order to check the SIP signalling for policy and charging purposes.

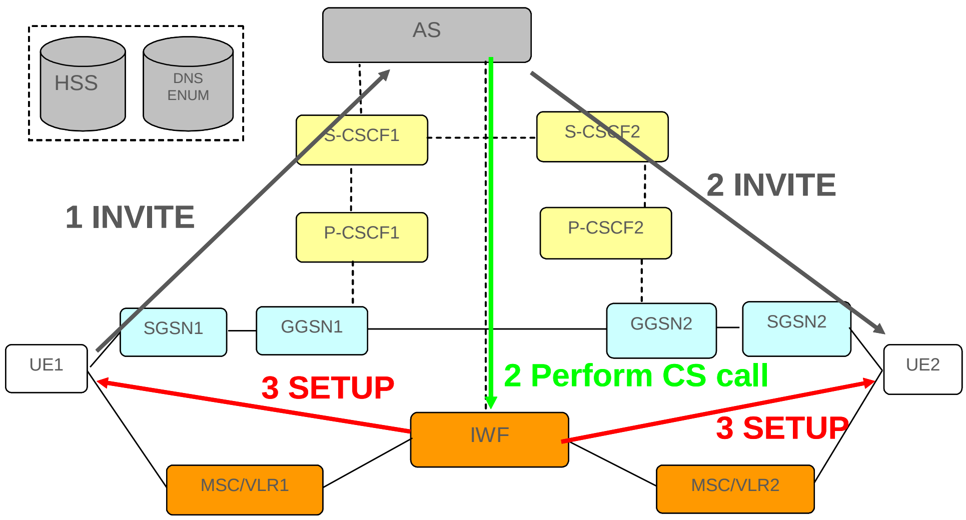

Figure 6.3.1-1 Architecture configuration overview

The basic steps necessary to perform an IMS session setup requiring a Circuit Bearer are described hereafter.

- Negotiation is based on the capability to indicate Circuit Bearers within a Media Component of a Session Description: this indication includes an E.164 address (possibly in the 'c=' line).

- The Originating client that is capable of supporting IMS/CSB for voice services specifies in the SDP its E.164, as indicated above.

- To route the Originating client SIP signalling to the HPLMN IMS/CSB AS, a PSI introduced in the Request-URI can be used. Or if UE2's tel URI is used in the Request-URI, it may be used for initial filter criteria in the S-CSCF of UE1 with other part of the INVITE message in order to route the SIP signalling.
- The IMS/CSB AS residing in UE1 HPLMN routes SIP signalling to UE2 HPLMN S-CSCF. 3rd PCC will be performed by the IMS/CSB AS, on receipt of the SIP answer from the UE2 client. IMS/CSB can decide from the UE1 and UE2 SDP whether a CS bearer is needed or not.
- If a CS bearer is needed, the IMS/CSB AS performs a 3rd Party Call Control towards the two CS endpoints of the IMS session (UE1 and UE2). E.164 number for UE1 is derived by SDP 'c=' line; UE2 E.164 can be derived: by the tel:URI included in SIP Signalling itself, if any; or by IMS/CSB AS interrogations to DNS/ENUM database; or by the SDP 'c=' line introduced in the 183 Session Progress by the UE2.
- An Interworking Function in the network will map the IMS/CSB AS 3rd PCC into two CS Call Setup (network to client direction), which will be through-connected at the MGW.
- on receipt of the SIP INVITE generated from the IMS/CSB AS 3rd PCC, the IWF will setup the two CS calls towards the UE1 and UE2 using the exchanged E.164 numbers as follows:

  SETUP towards UE1: Calling Party Number = UE2 E.164
  SETUP towards UE2: Calling Party Number = UE1 E.164

  The UEs will use these information in order to correlate the CS and IMS bearers.
- In the following, we assume that the IWF is a MGCF and that the signaling between IMS/CSB and the IWF is SIP.
- Traditional ISUP CS signalling is used as bearer control protocol to establish a CS call towards each session end point.

Note that:

- Only one IMS/CSB AS need to be involved in the SIP signalling path to invoke a CS connection. Also a mechanism to ensure that terminating PLMNs' AS do not invoke itself a 3rd PCC is to be defined (e.g. on the base of Request-URI value of the SIP INVITE and the CSB indication in the SDP).
- Note that no AS needs to be invoked in UE2 HPLMN for specific IMS/CSB purposes.The terminating PLMNs' AS can still be invoked for other ISC interactions (e.g. policy,…). E.g. the terminating IMS PLMN will still have to be able to police IMS session initiations and modifications, in case it does not accept the CSB SDP.
- Differently from Alternative A, SDP is not tampered with by any CSCF or AS.
- In line with RFC 3725 "Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)", the IMS/CSB AS originates two sessions and acts as a central point for signalling, having complete control over the SIP session and over the CS bearer legs. The IMS/CSB AS is intended to be always "aware" of the service provided, i.e. the AS can handle IWF reports on the events which are occurring on the CS domain.
- The solution does not require modification to the CS domain nor to the CSCF/MGCF.

6.3.2 Example Call flows

This section presents two example call flows based on the above mechanism. For simplicity, not all IMS and CS domain nodes are shown. Signalling flows relevant to the trigger of CS bearer are coloured differently from traditional IMS Signalling.

All the call flows are referred to a general inter-operator case; however only interactions with IMS/CSB AS are shown.

6.3.2.1 Voice-only IMS session set up

In Figure 6.3.2.1-1, UE1 sends an INVITE message specifying only the media audio in the SDP, and indicating the capability to handle Circuit Bearers. The S-CSCF in HPLMN recognizes that an IMS/CSB AS for Combining CS bearer is needed. An IWF is triggered by the IMS/CSB AS in order to establish two 24.008/ISUP MT calls towards UE1 and UE2 in 3PCC.

The overall session control is kept at IMS level. The IMS/CSB AS can be always aware of the events which are occurring on the CS domain thanks to IWF reports.

An upgrade of the session to eventually add a PS component can be performed using standard IMS signaling, as for normal procedure in TS 23.228.

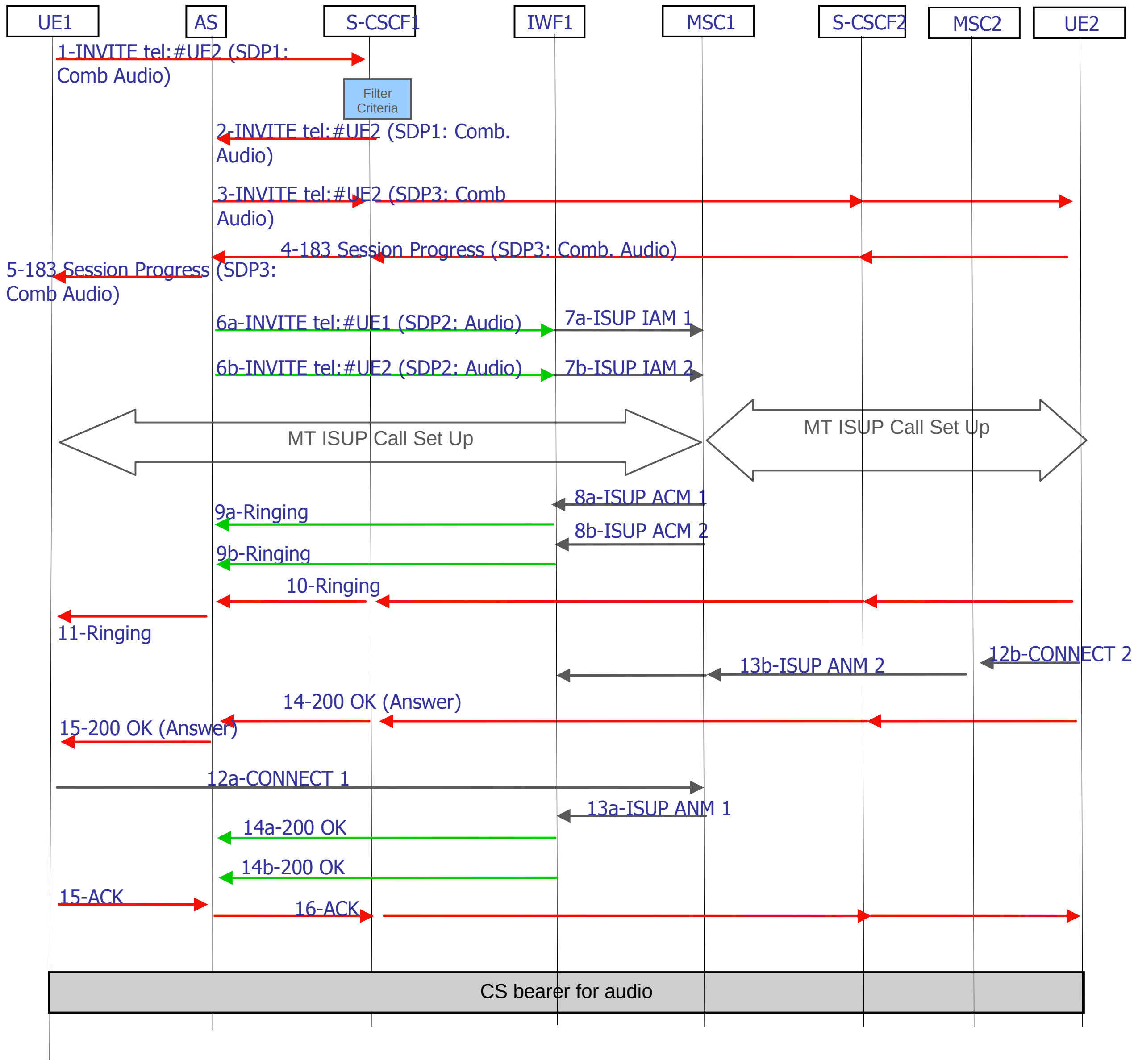

Figure 6.3.2.1-1 Example call flows for voice-only IMS Session Set up

6.3.2.2 Multimedia (voice + data) IMS session set up

In 6.3.2.2-1, UE1 sends an INVITE message specifying audio and another generic media in the SDP. For the media audio the capability to handle Circuit Bearers is indicated. The S-CSCF in HPLMN recognizes that an IMS/CSB AS for Combining CS bearer is needed.

An IWF is triggered by the IMS/CSB AS in order to establish two 24.008/ISUP MT calls towards UE1 and UE2 in 3PCC.

The overall session control is kept at IMS level. The IMS/CSB AS can be always aware of the events which are occurring on the CS domain thanks to IWF reports.

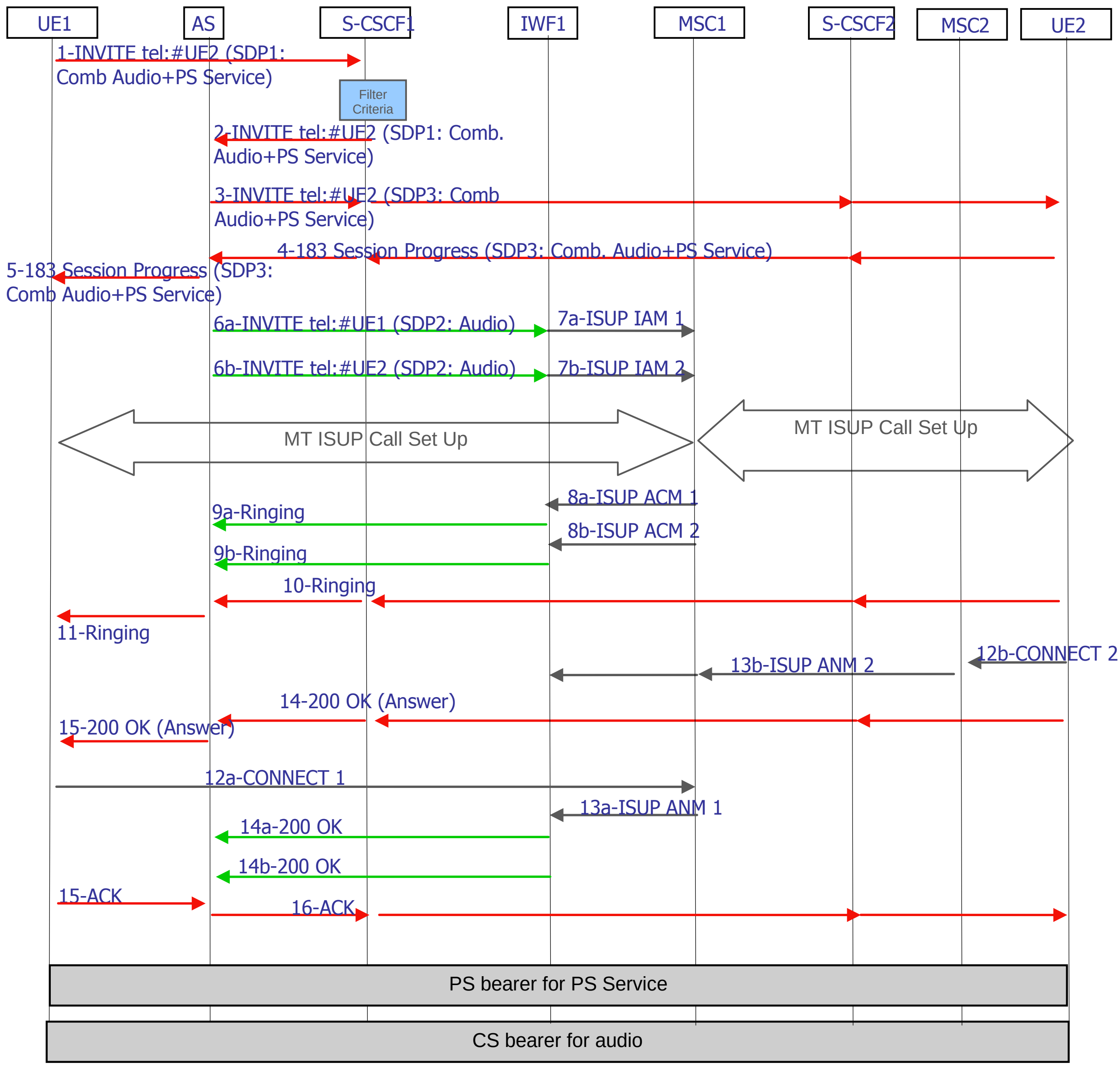

Figure 6.3.2.2-1 Example call flows for a generic IMS Session Set up including voice media

6.3.3 Considerations on interaction with CS Supplementary Services

Alternative C foresees that the CS bearer combined to an IMS session is established in a network-to-client direction (the call is established both at UE1 and UE2 side as a terminating CS call).

Outgoing call barring supplementary services and outgoing CAMEL services are not affected by IMS/CSB logic. Only interaction of CSB logic with some CS terminating services needs to be addressed.

To manage interaction with some CS Supplementary (terminating) Services, in general two cases can be distinguished:

1. CS & IMS call forwarding settings are aligned
2. CS & IMS call forwarding settings are not aligned

Case 1

In case 1, any call forwarding setting and UE status (e.g. CS busy) can be generally detected at SIP level before than CS level, and thus it can be handled by terminating IMS according to the desired logic without any need for triggering the CS corresponding logic.

Case 2

In case 2, the IMS/CSB logic has to cope with whatever action the terminating CS domain performs with regard to terminating CS services. This means that any event occurring on the CS domain (eg. as a result of the activation of a CFB) needs to be notified back to the IMS/CSB AS in charge of 3rd PCC, which in turns will handle the IMS session set up accordingly.

A specific case of interaction of Alternative C procedures with CS Call Forward on Busy (CFB) at UE2 side is described as an example (Figure 6.3.3-1).

- UE1 invites UE2 to an IMS session, composed of a voice (indicating IMS/CSB capability in the SDP) and a generic data media.
- UE2 has activated a CS CFB, but not the corresponding IMS CFB.
- IMS/CSB AS in PLMN1 invokes 3rd PCC and triggers a CS set up towards UE1 and UE2.
- As a result of CFB, CS call to UE2 is redirected towards UE3. MSC2 notifies back to IWF this event, including UE3 MSISDN (using standard ISUP signalling).
- IWF notifies this event to IMS/CSB AS, including UE3 MSISDN.
- IMS/CSB in turn closes IMS session towards UE2 and initiates a new IMS session towards UE3.
- UE3 will send a CS Connect only on receipt of the SIP ACK from UE1.

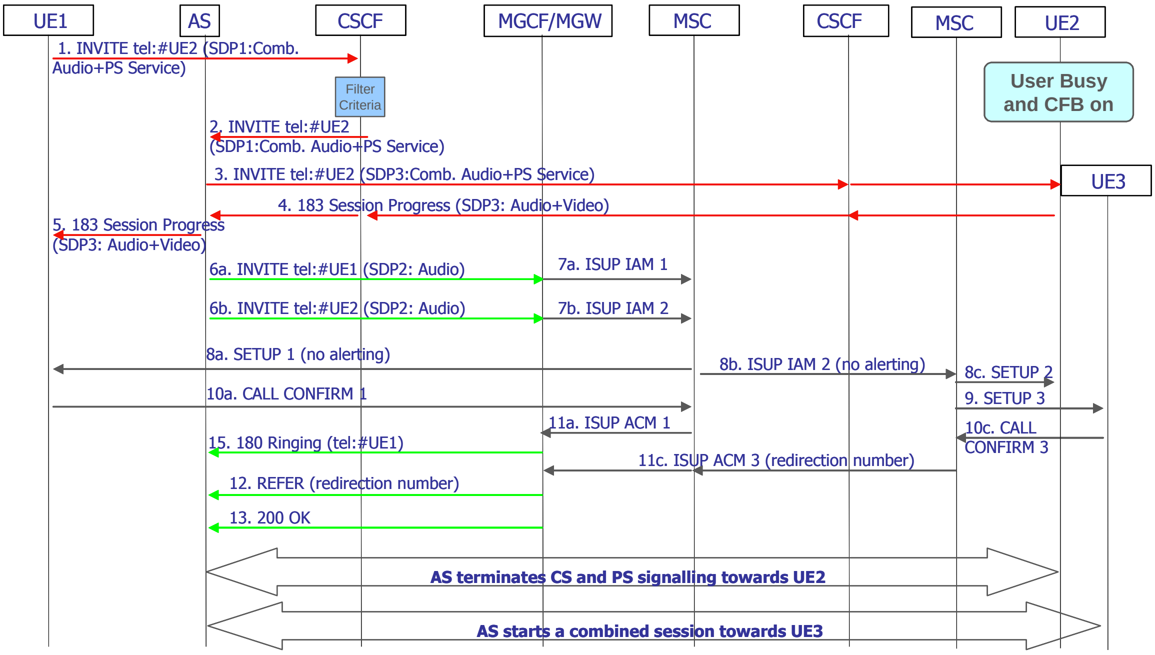

Figure 6.3.3-1 Example call flows for a generic IMS Session Set up towards a user with CS CFB active

The following remarks apply:

- The IMS/CSB AS is always aware of the events which are occurring on the CS domain.
- On CS domain, terminating services are handled in the traditional way.
- The IWF reports actions performed by CS domain on terminating services back to IMS/CSB AS.
- IMS/CSB AS handles SIP session accordingly (e.s. initiates a new session towards the CS redirected destination).

Based on the previous considerations, the case 1 that foresees alignment of CS and IMS call forwarding represents the simplest solution to handle interaction with supplementary service.

6.4 Alternative D

6.4.1 Exchange of capability information “at” CS call setup

Exchanging terminal capabilities and user preferences for combinational services is useless or impossible if either party doesn’t currently have access to the required radio capabilities. As an optional optimisation, the following end-to-end radio capability information exchange procedure in CS signalling may be executed prior to the SIP based terminal capabilities information exchange procedure. The radio capability information exchange procedure may indicate that a combinational service is not possible, but the combinational services as such shall not be affected by the execution or result of this optional procedure. The use of the radio capability information exchange procedure is particularly recommended, if UE-A has cached information on UE-Bs terminal capabilities.

Note: There will exist UEs, which do not support the radio capability exchange procedure, but do support parallel CS calls and IMS sessions, e.g. Rel-5 IMS-capable UMTS UEs. Thus lack of an answer in the radio capability exchange procedure does not mean that the remote UE cannot handle a parallel IMS session or the SIP based capability exchange.

The first sequence diagram outlines the exchange of current radio capabilities, e.g. “DTM cell”, “at” CS call setup. The key for this capability exchange is the use of, e.g., the “subaddress” fields or user-to-user signalling (UUS) to encode the information; see section 7.1. for examples of this information. The diagram shows only an example of actual messages that can be used to transport this information. User-to-user signalling is specified in TS 23.087 [7]. For this procedure to be successful, the “subaddress” fields or UUS must be handled transparently by the network. In case of UUS UUS Service 1 would be used, which allows the transfer of user-to-user information embedded within call control messages.

Editor’s note: the use of subaddress or UUS-1 (or another end-to-end-mechanism) to transport the current radio capabilities is ffs.

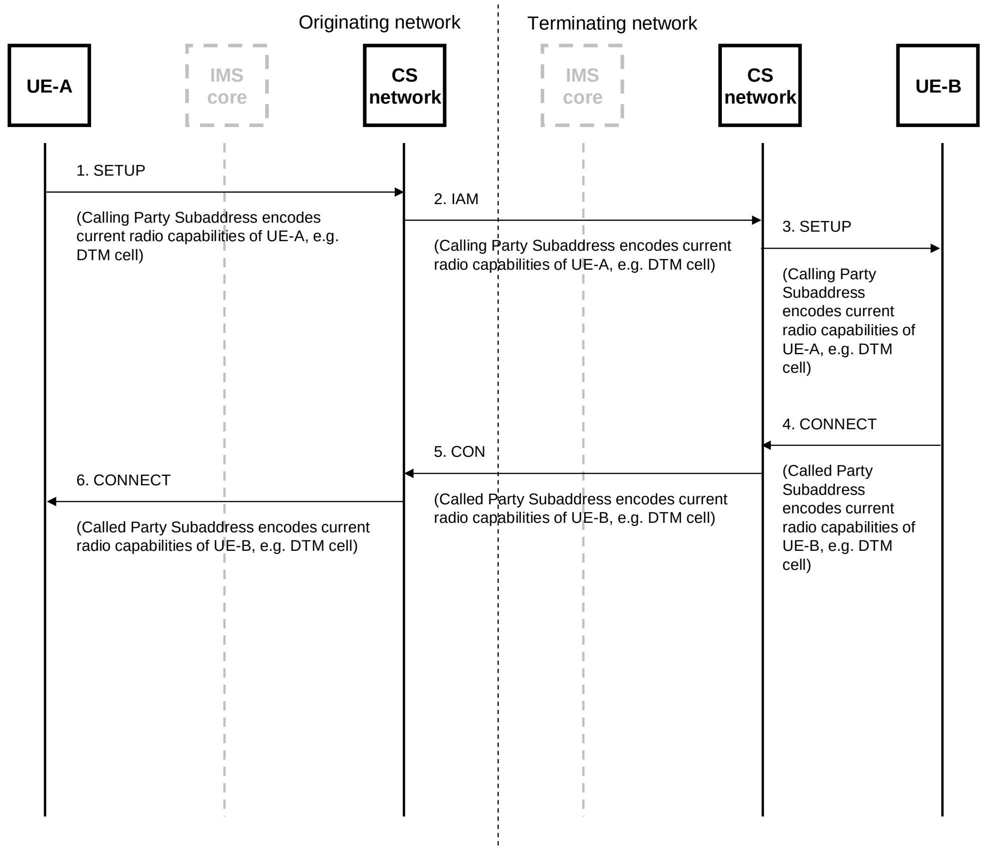

Figure 6.4.1-1: Exchange of current radio capability information “at” CS call setup

The following sequence diagram outlines the exchange of terminal capability and user preference information “at” CS call setup. The use of the SIP OPTIONS request minimizes the amount of network signalling and resource usage as well as the number of failed INVITE requests. . It also allows an up-to-date indication to the user which capabilities he could add to the ongoing call.

The execution of this OPTIONS request procedure is RECOMMENDED when UE-A's cache doesn't contain up-to-date information for UE-B.

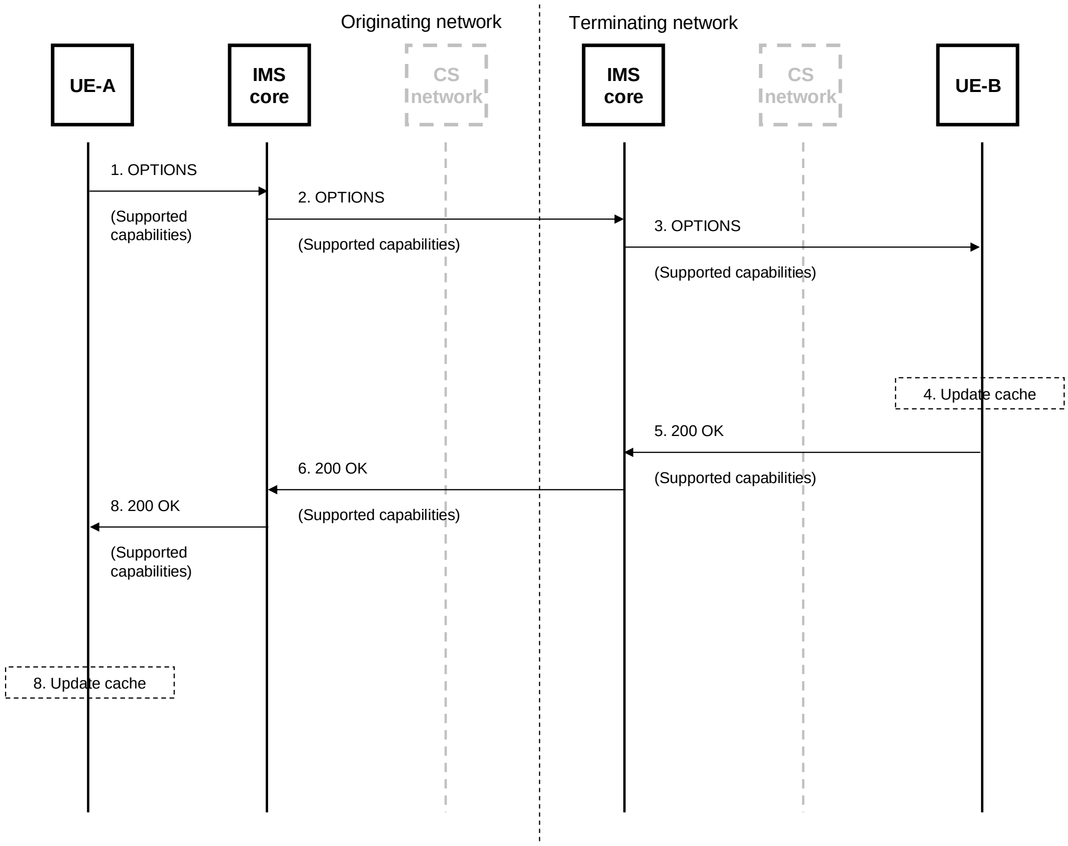

Figure 6.4.1-2: Exchange of supported capabilities "at" CS call set-up

Note: See Section 7 for considerations on the SIP message to be used for this exchange.

In this scenario it is assumed that UE-A either has information (e.g. in the phone book) regarding the SIP-URI of UE-B, or that it can use an address based on the E.164 number used for the CS call, e.g. Tel-URI.

6.4.2 Exchange of capability information "at" IMS session setup

In this case, the standard IMS session setup exchange is enhanced with capability exchange information, including the exchange of E.164 numbers for the possible addition of a CS call later in the session.

The inclusion of the E.164 number by the UE indicates that it is prepared to accept an incoming CS call in combination with this IMS session and also indicates the calling line identity that will be used if the UE later initiates a CS call in combination with this IMS session. It is expected that such mutual exchange of E.164 numbers (to be used for peer-to-peer CS voice calls between the same users) within SIP session establishment requires new functionality either on the SIP or the SDP protocol level. Hence, it is assumed that this functionality (along with functions building on this capability) will not be available in the initial phase of combinational services launch.

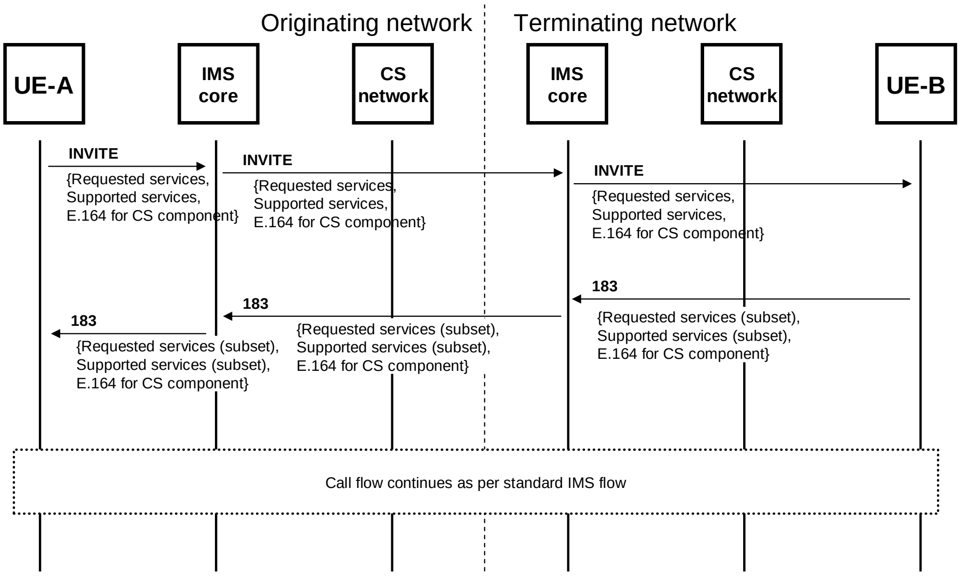

Figure 6.4.2-1: Exchange of capability information at IMS session establishment

Note: other IMS elements (CSCFs, ASs, BGCFs etc.) and other CS domain elements (G-MSC etc.) are not shown for simplicity.

The possibility to exchange radio capabilities in this flow should also be considered. See Section 7.

6.4.3 Customer adds an actual IMS service to an ongoing CS call

The following sequence diagram shows an IMS service being added to an ongoing CS calls:

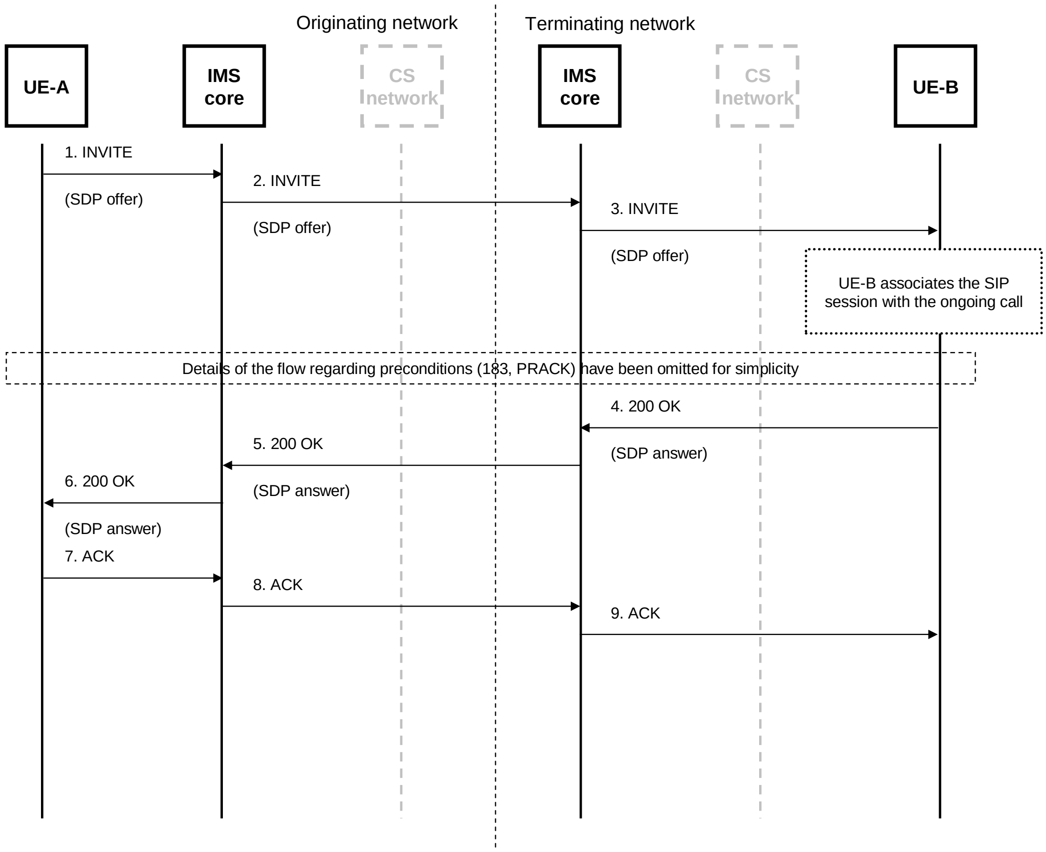

Figure 6.4.3-1: UE-A adds an IMS service to an ongoing CS call

In this scenario it is assumed that UE-A either has information (e.g. in the phone book) regarding the SIP-URI of UE-B, or that it can use an address based on the E.164 number used for the CS call, e.g. Tel-URI. If the same E.164 number is used in IMS and the CS domain, it shall be possible to deliver both the IMS session and the call destined to the E.164 number of the UE, and the call must stay within the domain it starts in.

6.4.4 Customer adds a CS call to an ongoing IMS session

The E.164 called party number to be used for adding a CS service to an ongoing IMS service may be *a priori* available, e.g. in the phone book of the users' terminal, or if the same E.164 number is used in the CS domain and in the IMS. It may also be retrieved via means that are outside the scope of 3GPP standards (e.g. exchanged via email). Alternatively, the E.164 number to be used for adding a CS service may be negotiated during IMS session setup as per Section 6.4.2. This flow may occur at any point after E.164 numbers have been exchanged according to Section 6.4.2. In particular, if UE-A knows that the users wishes to establish a session with a voice component, rather than a non-real-time session to which voice may be added later, then this flow will be initiated immediately on reception of the E.164 number from UE-B.

The flow below can apply in either direction i.e. initiated by UE-A or UE-B.

Note, if is ffs whether the flow in 6.4.2 should be restricted to negotiating the capability to add a CS call. If so, another IMS message exchange may be required to negotiate the actual addition of the CS call.

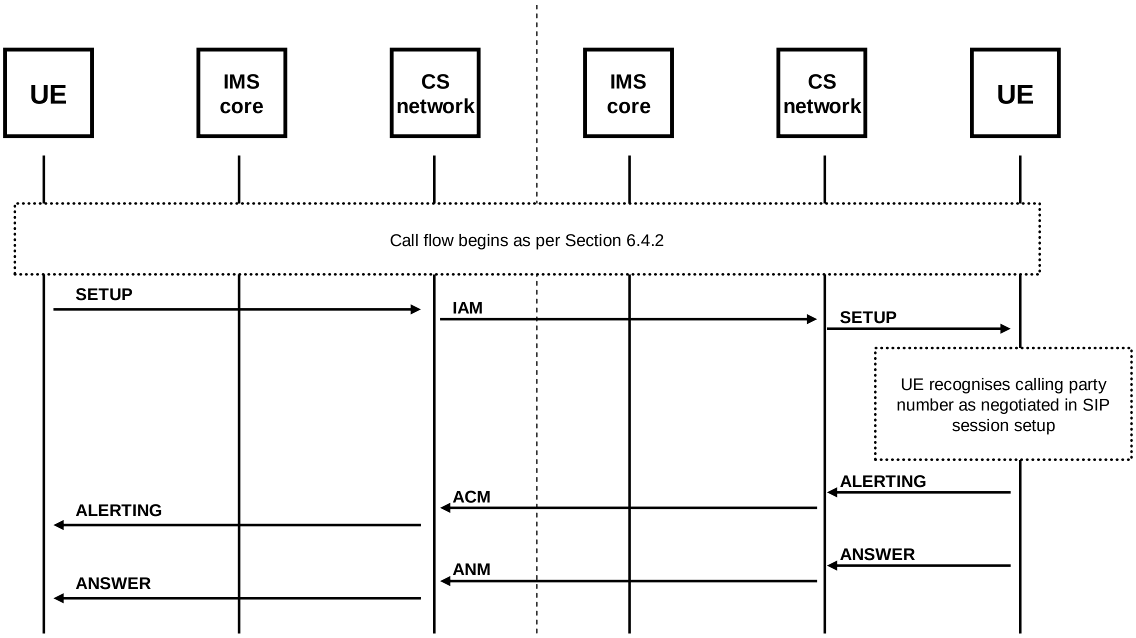

Figure 6.4.4-1: Customer adds a CS call to an ongoing IMS session

6.4.5 Multicomponent (including voice) call from pure VoIP end point

6.4.5.1 General

These sections demonstrate how the capabilities used for cases 6.4.1-4 can be applied to support calls from SIP endpoints supporting VoIP.

Editor's Note: other flows are ffs.

These flows show how multicomponent (including voice) IMS sessions originated from a pure VoIP A-party can be delivered to a B-party that uses CS bearers for the voice component. It is assumed that the pure VoIP terminal does not support the CSI mechanisms, hence the serving network of the B-party needs to provide an E.164 number to be used for the CS call. This number is then provided to the B-party using the negotiation mechanims described in clause 6.3.2. As the mechanism in 6.3.2 requires new functionality either on the SIP or the SDP protocol level, it is assumed that the functionality of delivering multicomponent IMS sessions from pure VoIP parties to CSB parties will not be available in the initial phase of combinational services launch.

Note: Both network-to-client and client-to-network call flows are shown below for completeness. Further work should identify a single option to be further discussed.

6.4.5.2 Network-to-client CS call

This flow uses the E.164 number negotiation of Section 6.4.2 to inform the terminating UE of the calling line identity that will identify a forthcoming incoming CS call and for the terminating UE to indicate its willingness to accept such a call. This CS call will be interworked to the VoIP component from the pure VoIP endpoint.

The terminating UE procedures are as per 6.4.2 and 6.4.4 above.

Note that in the case that simultaneous CS and PS connectivity is not available at UE-B, then further SIP signalling cannot be exchanged after the SETUP message has arrived at UE-B. Any IMS media components without pre-conditions will remain in place, although no data will flow. IMS Media Components with pre-conditions will never be established.

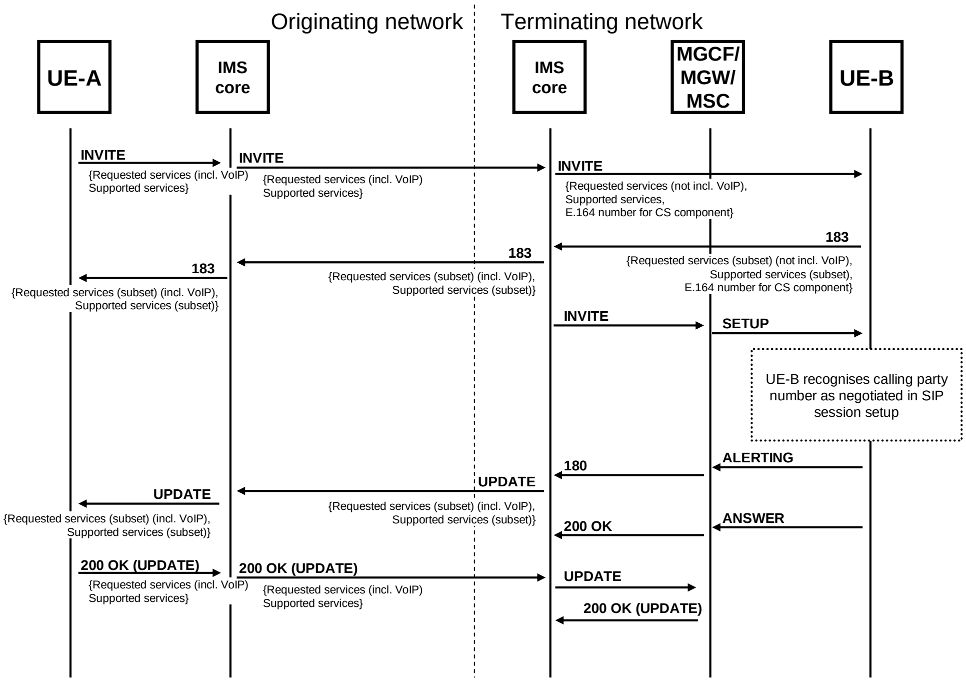

Figure 6.4.5.2-1: Call from pure VoIP endpoint using network-to-client CS call

Alternatively, the CS call may be established first, with the IMS session being established after the CS call is in the Alerting state. In this case, the terminating UE procedures would be the same as 6.4.1 followed by 6.4.3 above. The end result is the same. If simultaneous PS and CS connection is not available in this case, the IMS session will fail altogether.

Note that if capabilities are defined to indicate simultaneous CS and PS connectivity within CS domain signalling, this information could be made available to the Application Server controlling the delivery of the call.

6.4.5.3 Client-to-network CS call

This flow uses the E.164 number negotiation of Section 6.4.2 and an additional indication to request the terminating UE to establish a CS call to the provided number. This will then be interworked to the VoIP component from the pure VoIP endpoint.

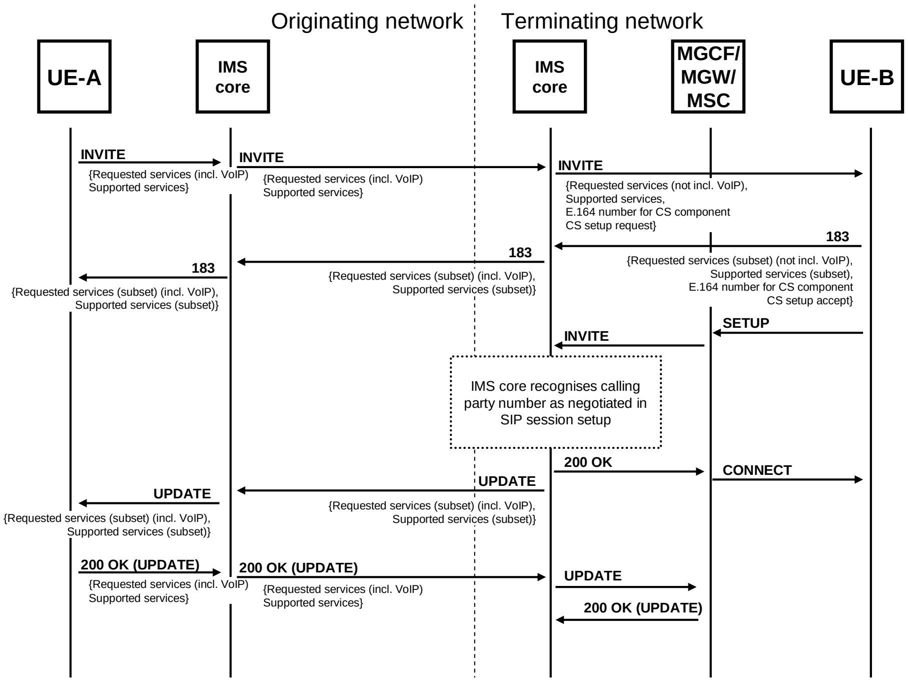

Figure 6.4.5.3-1: Call from pure VoIP endpoint using client-to-network CS call

6.4.6 Multicomponent (including voice) call to pure VoIP end point

6.4.6.1 General

These sections demonstrate how the capabilities used for cases 6.4.1-4 can be applied to support calls from SIP endpoints supporting VoIP.

Editor's Note: other flows are ffs.

These flows show how multicomponent (including voice) IMS sessions originated from an A-party that uses CS bearers for the voice component can be delivered to a pure VoIP B-party. It is assumed that the pure VoIP terminal does not support the CSI mechanism, hence the serving network of the A-party needs to "hide" the usage of CS bearers from the B-party. This uses the E.164 negotiation mechanism described in clause 6.4.2 in the A-party's network. As the mechanism in 6.4.2 requires new functionality either on the SIP or the SDP protocol level, it is assumed that the functionality of delivering multicomponent IMS sessions from CSB parties to pure VoIP parties will not be available in the initial phase of combinational services launch.

Note: Both network-to-client and client-to-network call flows are shown below for completeness. Further work should identify a single option to be further discussed. This may or may not be the same option as chosen for calls from a VoIP endpoint.

6.4.6.2 Network-to-client CS call

This flow uses the E.164 number negotiation of Section 6.4.2 for the UE to inform the network that it will accept an incoming CS call and for the network to inform the UE of the calling party number that will identify this call.

It is assumed that UE-A knows whether the user is attempting to initiate a call with a voice component, or just a non-real-time IMS session to which voice may be added later. In the former case, UE-A explicitly requests the network to establish the CS call. The CS call will be interworked to the VoIP component from the pure VoIP endpoint.

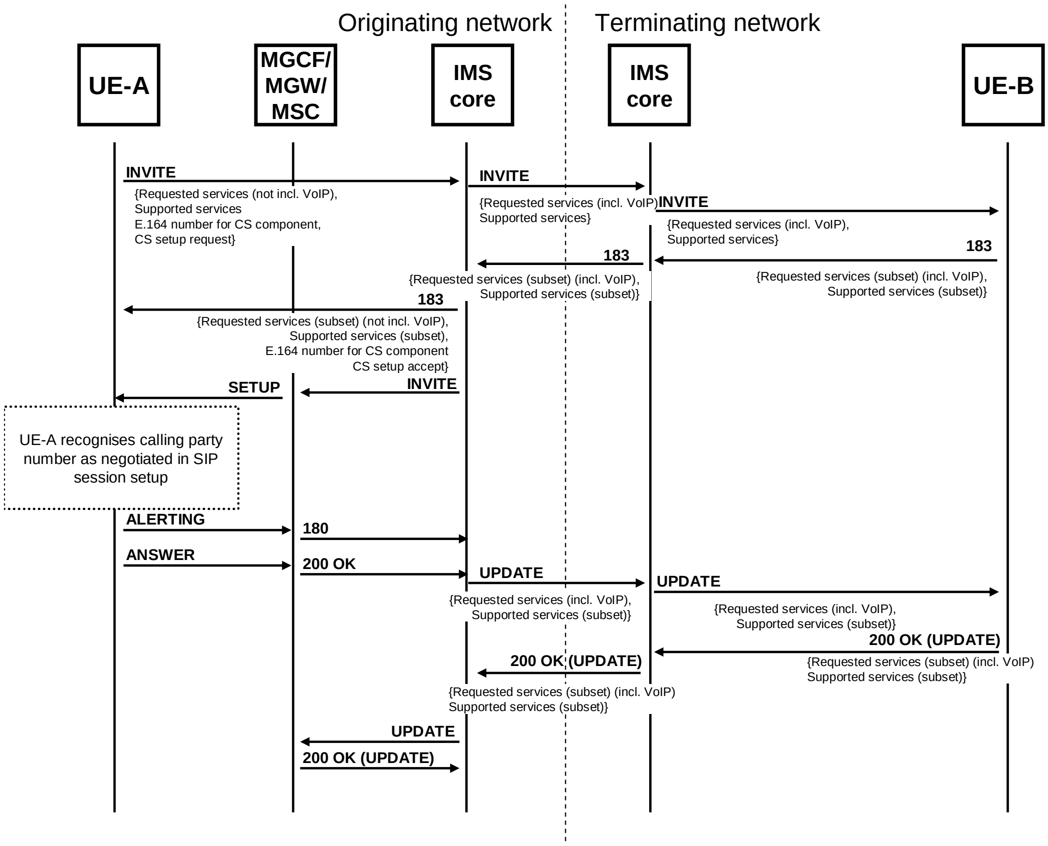

Figure 6.4.6.2-1: Call to pure VoIP endpoint using network-to-client CS call

6.4.6.3 Client-to-network CS call

This flow uses the E.164 number negotiation of Section 6.4.2 for the UE to inform the network that it may establish a CS component and for the network to inform the UE that it will accept such a call. The voice call will be interworked to the VoIP component from the pure VoIP endpoint.

The originating UE procedures are identical to 6.4.2 followed by 6.4.4 above.

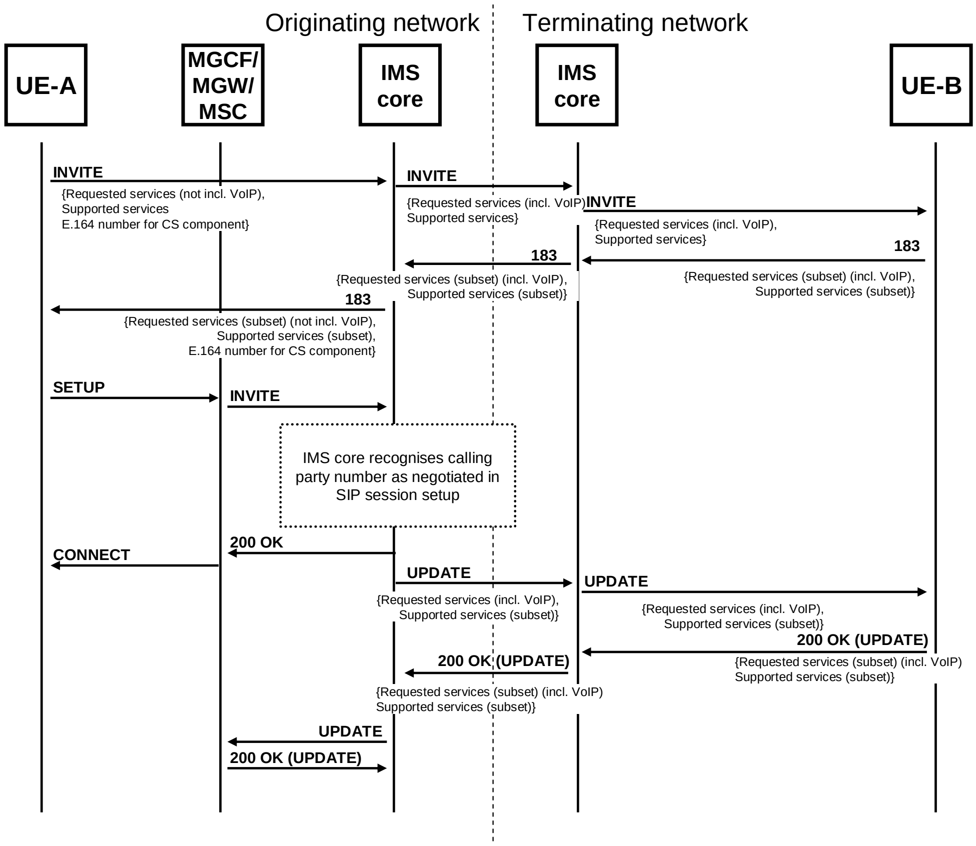

Figure 6.4.6.3-1: Call to pure VoIP endpoint using client-to-network CS call

Alternatively, if the VoIP endpoint has an E.164 number, the originating UE may originate the CS call first, followed by the IMS session. In this case the originating UE procedures are identical to 6.4.1 followed by 6.4.3 above.

However, the terminating VoIP endpoint will not support the CSI procedures and hence will not respond to the E.164 CS call numbers within the OPTIONS or INVITE exchange. This informs the originating UE that the two services, CS and IMS, will be handled independently by the terminating UE. The originating UE may re-originate the session using the procedures above to obtain a true 'combinational' service.

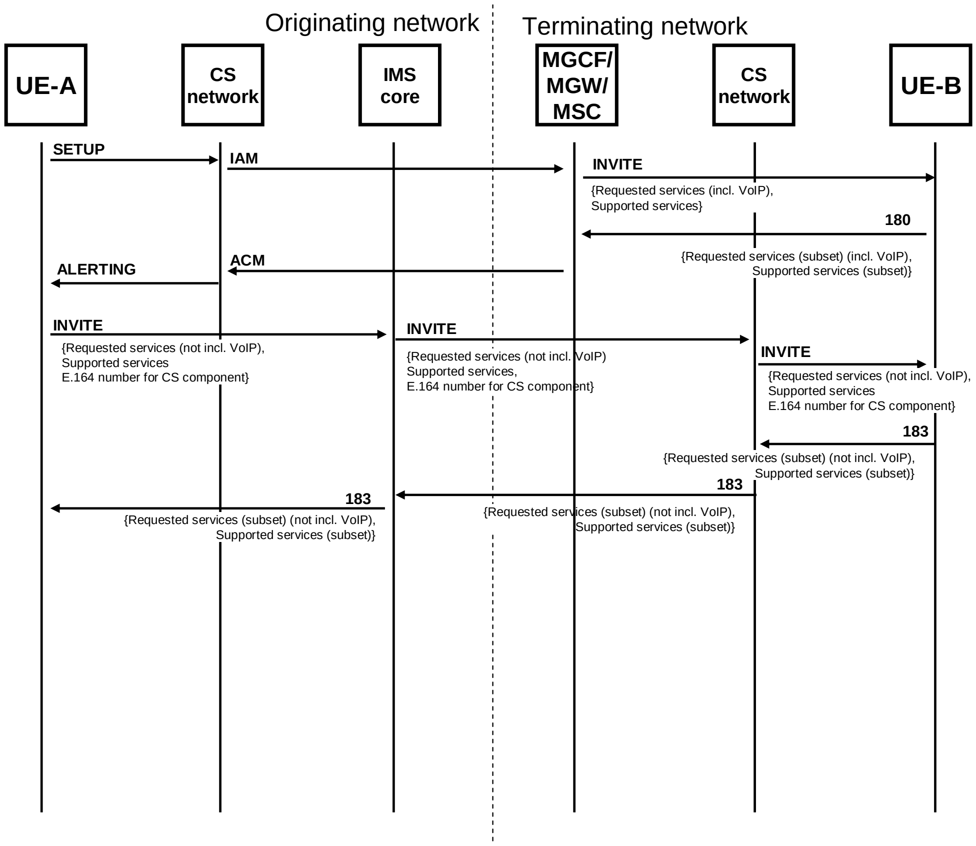

Figure 6.4.6.3-2: Independent IMS session and CS call to pure VoIP endpoint

6.4.7 Originating network control

6.4.7.1 General

This section demonstrates how the capabilities of Section 6.4.1 to 6.4.4 can be applied to create a CS connection between two parties to an IMS session under control of the originating network.

This is achieved using the procedures of 6.4.6 at the originating side to connect a CS call between the originating user and a gateway. The procedures of 6.4.5 are then applied under originating network control to connect a CS call between the terminating user and the same gateway.

As with the referenced sections, both network-to-client and client-to-network cases are described.

6.4.7.2 Network-to-client CS call

This flow uses the E.164 number negotiation of Section 6.4.2 for the UE to inform the network that it will accept an incoming CS call and for the network to inform the UE of the calling party number that will identify this call.

It is assumed that UE-A knows whether the user is attempting to intiate a call with a voice component, or just a non-real-time IMS session to which voice may be added later. In the former case, UE-A explicitly requests the network to establish the CS call.

The client procedures at the terminating UE are identical to 6.4..4 above

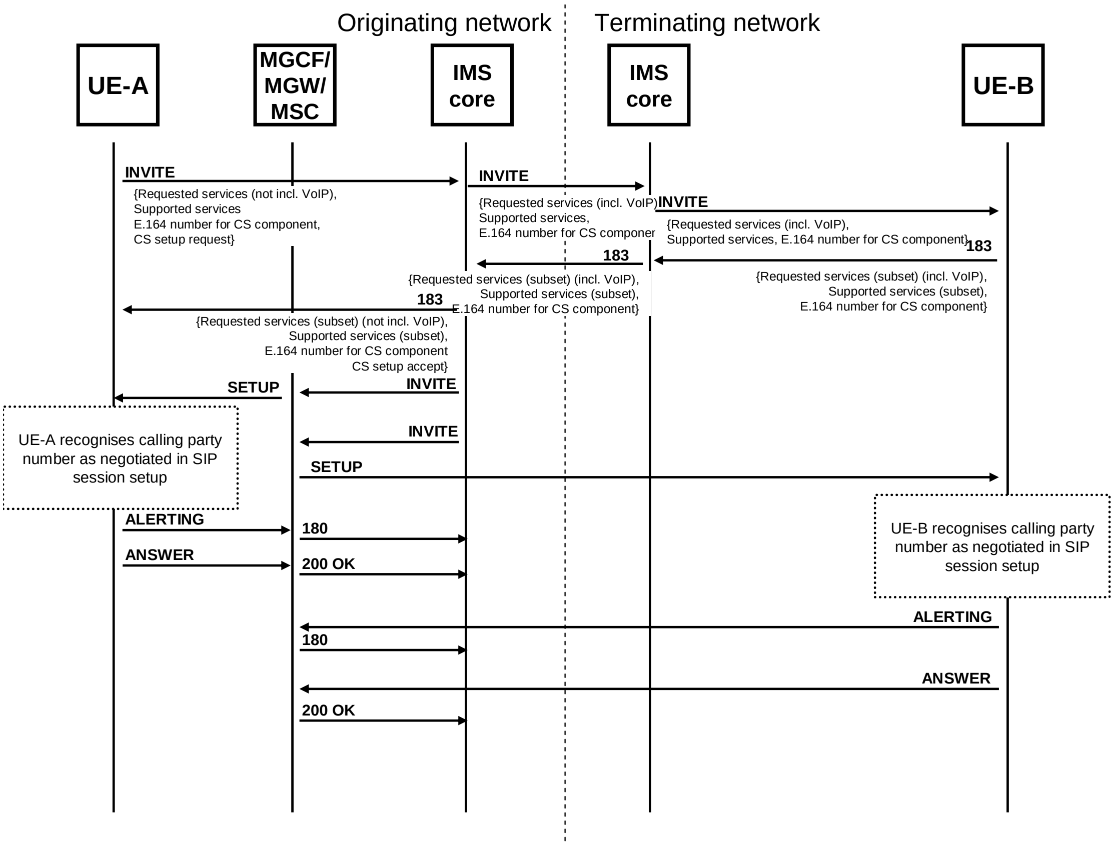

Figure 6.4.7-1: Network to client calls with originating network control

6.4.7.3 Client-to-network CS call

This flow uses the E.164 number negotiation of Section 6.3.2 for the UE to inform the network that it may establish a CS component and for the network to inform the UE that it will accept such a call. The voice call will be interworked to the VoIP component from the pure VoIP endpoint.

The originating UE procedures are identical to 6.4.2 followed by 6.4.4 above.

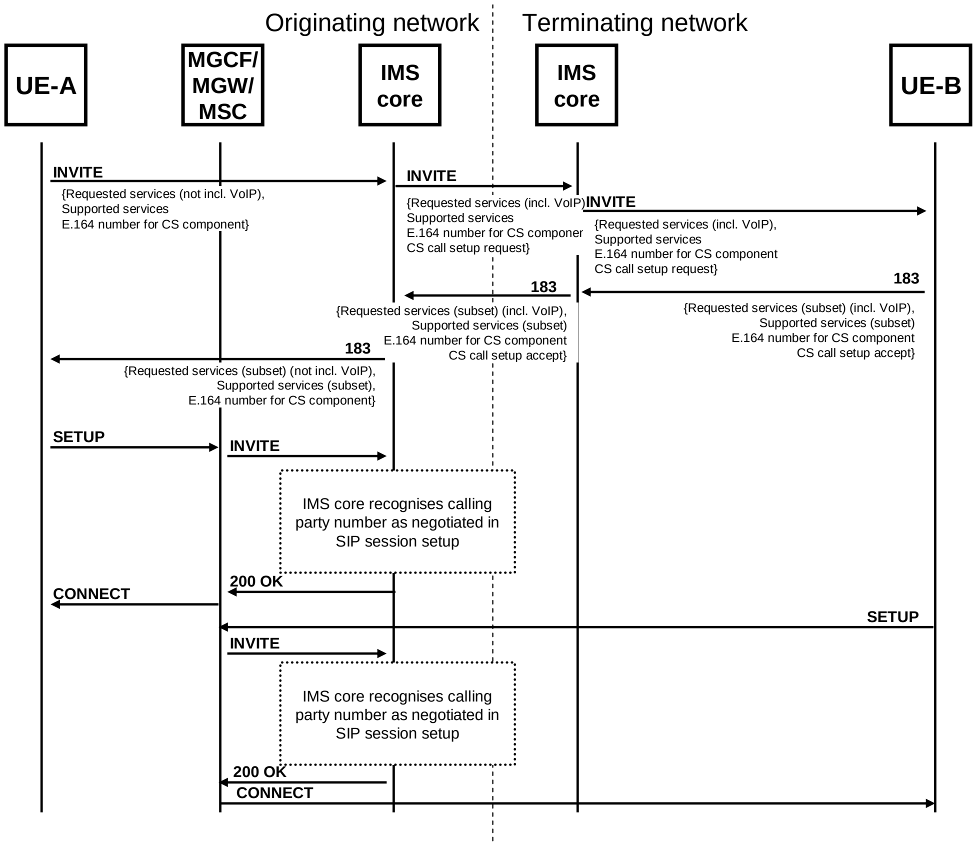

Figure 6.4.7-2: Client to network calls with originating network control.

6.4.8 Client control Multimedia call interworking

6.4.8.1 Multimedia call from pure VoIP end point

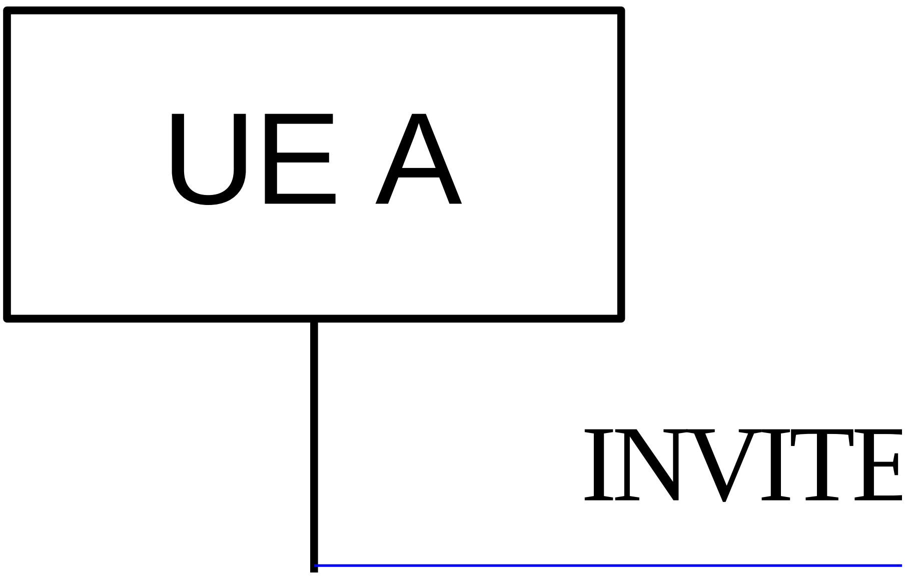

Figure 6.4.8.1-1: Multimedia call from pure VoIP end point with client control

This flow show how multimedia (including voice) IMS sessions originated from a pure VoIP A-party can be delivered to a B-party that uses CS bearers for the voice component.

- Since the UE A is a pure VoIP UE and does not support CSI function, so the INVITE message sent by UE A is a normal one without CSI indication.
- CSI UE B knows that UE A doesn't use CSI capability, so decides to trigger embedded CSI service logic in end-to-gateway mode. It then originates a CS call while the calling party number is its MSISDN and the called party number is the E.164 number used for its IMS part (may be configured by adding a prefix to its MSISDN).

Note: the E.164 number included in the SETUP message is constructed by adding a prefix to its MSISDN, the prefix is to be predefined for CSI services by the operator B (i.e. it is network B specific). The UE needs to modify the MSISDN to include this prefix when it initiates a CS call with a VoIP peer. Alternatively, prefix could be added to the MSISDN by the user.

- The CS domain routes the CS call to an CS/IMS interworking gateway based on its configured routing data, the special prefix for CSI can be used by CS domain to choose the nearest interworking point or/and to avoid the impact of CS originating side supplementary service (just use the same method having been adopted in the CS domain to avoid the impact of originating side supplementary service to the emergency service and special number service).
- The interworking gateway (MGCF/MGW) performs the CS/IMS interworking and routes the IMS session via IMS network to UE B's IMS part, bring the parameters of the port reserved for IMS side in MGW (SDP of GW for VoIP).
- UE B receives the INVITE message and distinguishs that it is a CSI call-back by founding it is from the MSISDN of itself, then control in B2BUA mode to perform exchange of SDP for VoIP component between UE A and MGCF/MGW, and so controls UE A exchange the media of VoIP component with MGW via IP bearer, then this VoIP component can exchange between MGW and UE B via CS bearer.
- To avoid UE A send media of VoIP component before the CS bearer establish, the attribute of SDP in 183 sent to UE A should be inactive, and when finished the establishment of CS bearer, UE B will send a UPDATE to change the attribute to active.

6.4.8.2 Multimedia call to pure VoIP end point

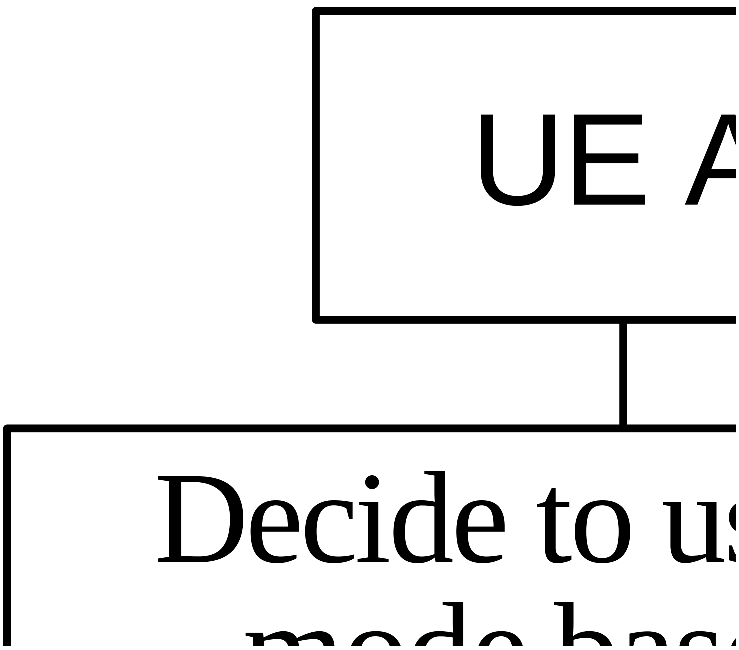

Figure 6.4.8.2-1: Multimedia call to pure VoIP end point with client control

This flow show how multimedia (including voice) IMS sessions originated from an A-party that uses CS bearers for the voice component can be delivered to a pure VoIP B-party.

The procedure is similar to "Multimedia call from pure VoIP end point (Client-to-network, client control)" excepting that, UE A may decide to trigger embedded CSI service logic in end-to-gateway mode based on capability exchange and can originate CS call back to itself to get the parameters of GW before it initial the IMS session.

Note: By some means, UE A is aware that UE-B is an IMS terminal. The E.164 number included in the SETUP message is constructed by adding a prefix to its own MSISDN, the prefix is to be predefined for CSI services by the operator A (i.e. it is network A specific). The UE needs to modify the MSISDN to include this prefix when it initiates a CS call towards its IMS part to establish the CS bearer to the GW and get the SDP of GW, which are then used in the subsequent IMS session it originates toward the VoIP peer. Alternatively, prefix could be added to the MSISDN by the user.

6.4.9 Call clearing after handover into non-DTM and non-3G area

Editor's note: ffs

6.4.9.1 Fallback to CS when access in non-DTM and non-3G area

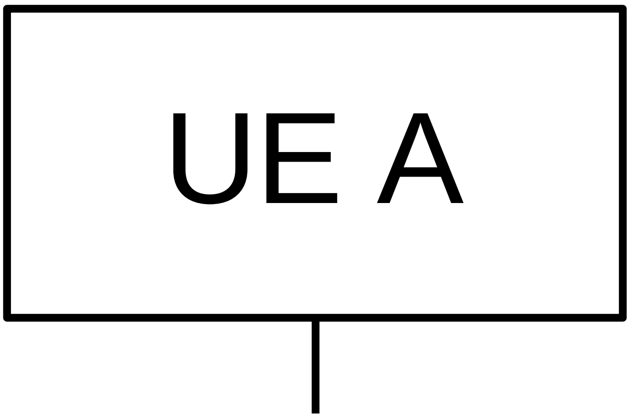

Figure 6.4.9.1-1: Fallback to CS when access in non-DTM and non-3G area

This flow show how multimedia (including voice) IMS sessions originated from a pure VoIP A-party be delivered to a B-party that uses CS bearers for the voice component, and then B-party decide to fallback to CS since it currently access in a non-DTM and non-3G area.

- When CSI UE B receives the INVITE message and triggers embedded CSI service logic in end-to-gateway mode, It then checks the current Radio access capability. Finding that the RAN it currently access doesn't support DTM/multiRAB, it then decides to fallback to CS and responds a 3xx to UE A including it's MSISDN.

- UE A responds ACK to confirm the 3xx and completes the release of the session being established, then originates a new INVITE to the MSISDN according to the 3xx.

- The IMS network A determines to transfer the newly originated session to CS domain based on the analysis of the destination address, and selects the interworking gateway. The interworking gateway (MGCF/MGW) performs the CS/IMS interworking including the media negotiation, and routes the CS call via CS network to UE B's CS part.

- UE B accepts the incoming CS call and then establishs a CSI fallback CS call with pure VoIP UE A. Being restricted by the capability of CS bearer, there may be a reduction in the exchanged media components if the session initially originated by UE A is a multimedia session, it is done by a standard media negotiation procedure between interworking gateway and UE A during the session establishment.

- Farther more, the E.164 number sent to UE A in 3xx may be configured by adding a prefix to UE B's MSISDN to force the re-originated session to be transfer to the CS domain, and interworking gateway is responsible to resume the MSISDN in this case.

6.4.10 Outline of proposed new SIP indications

6.4.10.1 General

The above call flows make use of three signalling elements which are proposed to be added to SIP to be included in SIP messages which carry an SDP offer or answer for an IMS session. Whether these are defined within the SDP message body, or the SIP message headers is a Stage 3 issue.

The CSI indication/CSI address defined below are used in the sections above to indicate that CSI is supported and the E.164 addresses that should be used.

The CSI request indication defined below is used in sections 6.4.5.3 and 6.4.6.2 to request immediate establishment of a CS call.

Related indications may also be required in SIP messages not associated with a session (e.g. OPTIONS). These are ffs.

6.4.10.2 Definitions

CSI indication:

This indication may be included in a SIP message associated with an IMS session. It indicates that a CS call, in either direction, may be associated with the IMS session.

The CS call may already be established, or may be established at a later time.

The CS call is identified by the "CSI address" parameter.

Note: the CSI indication need not be an explicit indication – for example it may be indicated implicitly by the presence of the CSI address. This is a Stage 3 issue.

CSI request:

This indicates that the receiver should establish a CS call in association with this IMS session.

The called party address for the CS call is provided in the "CSI address" parameter.

CSI address:

This parameter contains the E.164 number that will be used as called or calling party number for any CS call associated with this IMS session.

6.4.10.3 Client procedures

6.4.10.3.1 Outgoing IMS session

A CSI capable client may include the *CSI Indication* with any SDP offer that it sends. It shall also include its E.164 number within the *CSI address*.

On receipt of an SDP answer along with a *CSI indication*, the client shall:

- Check the received *CSI address* against any ongoing CS domain calls to determine whether the session should be immediately associated with the CS call
- If no matching call exists, the client shall store the received *CSI address*.
- Provide appropriate indications to the user about the available combinational services, based on the SDP answer.

Subsequently,

- if the user requests establishment of a CS domain call in association with the session, the stored *CSI address* shall be used.
- if an incoming CS call is received which matches the *CSI address*, the IMS session shall be associated with this call

On receipt of an SDP offer along with a *CSI request*, the client behaviour shall depend on whether the user requested a voice component to the session or not.

If a voice component was requested, then the client shall:

- Check the received *CSI address* against any ongoing CS domain calls.
    - o If a matching call exists, the session should be immediately associated with this CS call., otherwise
    - o if no matching call exists, the client shall establish a CS call to the provided *CSI address* and immediately associate the session with this CS call.

6.4.10.3.2 Incoming IMS session

On receipt of an IMS session request with the CSI indication or CSI request, the client shall follow the procedures of 6.4.x.3.1 with respect to receipt of an SDP offer.

7 Capability Exchange

7.0 General

It is highly advantageous if the set of services which can be supported between two endpoints is known to the endpoints when (or shortly after) communication is established. This information can be used to provide an indication to the user of the additional services which are available. This can encourage use of available services and avoid invocation of unavailable services, thereby avoiding customer dissatisfaction and unnecessary resource and bearer establishment attempts.

7.1 Terminal Capability Information

The information to be determined is the set of services that can be successfully invoked between two users at a given time. The information required for this can be divided in the following categories:

1. Terminal capabilities and user preferences:

   - CS voice call capability and the E.164 number used for voice calls
   - Media types which can be exchanged in SIP messages (i.e. MIME types that are acceptable in IMS Instant Message bodies);
   - Media types which can be supported as separate IMS media streams (i.e. media component definitions of IMS sessions);
   - Media format parameters for supported IMS media types (codecs, MIME types for Instant Messaging etc.);
   - CS video telephony capability;
   - MMS capabilities.

2. Radio network capabilities at the user's location:

   - Indication of simultaneous CS/PS service capability of the serving cell, i.e. DTM or UTRAN cell.

   Note: Exchanging such capabilities about the serving cell may allow the terminal to deduce certain service capabilities of the cell, e.g. UTRAN cell indicates support for CS video telephony capability.

   Radio network capability information is rather dynamic, hence it is assumed that such information would not be cached in the terminal.

   Radio network capability information is assumed to be application independent, e.g. GSM, DTM, UTRAN.

7.2 Alternatives

7.2.1 Alternative 1

7.2.1.1 Description of alternative

For a session in the CS domain, the network is responsible for triggering the request for capabilities (including terminal capabilities and serving network capabilities), determining the common capabilities and delivering this information to the endpoints participating in a session. In comparison, having the originating UE initiating capability exchange over IMS requires simultaneous communication in CS and IMS domain and may lead to increased signalling load especially when DTM is used to provide the communication mechanism. Increased signalling load arises from the request for DTM for delivery of individual messages and subsequent fallback to single transfer mode. Additionally, a serving BSS may not provide DTM capabilities and therefore any UE initiated mechanism may automatically fail (note: that handover to UMTS could be used by a "non-DTM" BSS to move the UE to UTRAN for any subsequent activity where simultaneous CS-voice and PS-data is required).

7.2.1.2 Mobile originating procedure

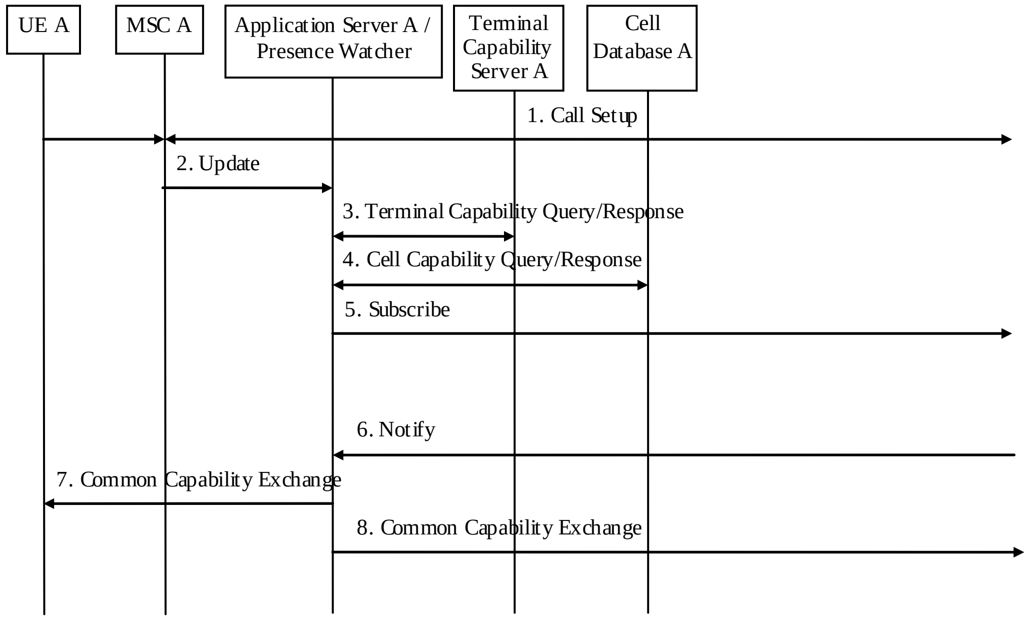

Figure 7.2.1-1: Mobile originating capability exchange procedure

1) UE-A initiates and progresses a CS voice call to a B-party.

2) At some point after or during call set-up the MSC notifies an application server (providing capability retrieval) of UE-A's serving cell id and terminal identification (e.g. IMEISV/UAProf) along with the event triggering this message.

3) From the terminal identification, the terminal capabilities can be determined, for example but not limited to video, MMS, UMTS capable.

4) From the serving cell id/service area id, the serving network capabilities can be determined, for example but not limited to DTM, EDGE, UMTS, HSDPA being locally available.

5) Whilst steps 3 and 4 are ongoing or afterwards, the application server, acting as a watcher, watches the presence information (capabilities and availability) of the called UE (UE-B) by sending a SUBSCRIBE.

6) The capability list and availability of UE-B is returned towards the application server of UE-A.

7) Some service logic is initiated to determine the common/shared capabilities and is reported to the UE-A using a pre-defined format. This may be delivered for example using SMS, network-initiated USSD or user-user signalling. The application server of UE-A may cache this information for future use.

8) The common/shared capabilities are reported to UE-B but may use a different transport mechanism (possibly determined by the determined capabilities) from the previous step.

Note: The terminal capability server and cell database are information stores which need not be separate entities accessed via new reference points. They are described as separate entities so that it is clear what information is used to obtain the capability information.

7.2.1.3 Mobile terminating procedure

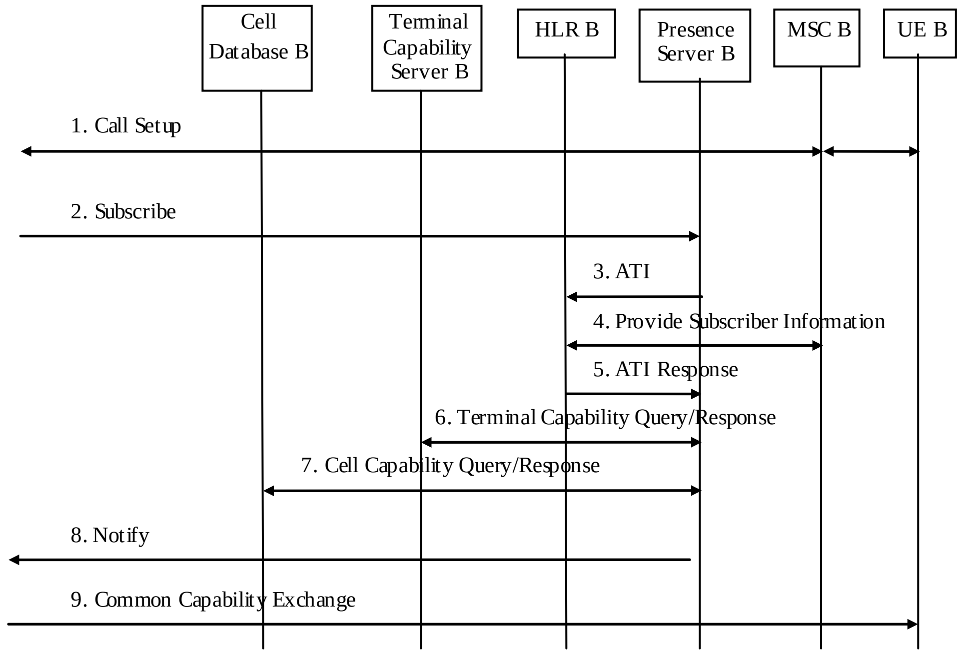

Figure 7.2.1-2: Mobile terminating capability exchange procedure

1) UE-A initiates and progresses a CS voice call to a B-party.

2) The presence server for UE-B receives a SUBSCRIBE from a watcher (application server of UE-A from section 7.2.1.2) to watch the capabilities and availability of the B-party (UE-B).

3) The Presence server serving UE-B queries the HLR/HSS of UE-B for the serving cell id/service area id and terminal identification(s) of UE-B.

4) The HLR in turn triggers a query towards the serving network.

5) The HLR returns the requested information to the Presence Server.

6) The presence server for UE-B queries for terminal capabilities using the obtained terminal identification(s).

7) The presence server then obtains the serving network capabilities based on the obtained cell id/service area id.

8) The capability list and availability is returned towards the watcher (application server of UE-A from section 7.2.1.2).

9) Once the processing in the originating network is complete, the common/shared capabilities are delivered to UE-B.

Note: The terminal capability server and cell database are information stores which need not be separate entities accessed via new reference points. They are described as separate entities so that it is clear what information is used to obtain the capability information.

7.2.2 Alternative 2

7.2.2.1 General

At any time (e.g., during an existing CS call, or prior to any other interaction with the remote UE), the UE can find out thecommon services that can be supported. The mechanism is based on SIP message exchanges with the remote UE. To reduce usage of radio bearer resources, the capability information may be cached in the UE for a predefined time-period.

It should be noted that caching of such information may result in the information becoming out-of-date. This could be for the following reasons:

- The device reachable through a given SIP Address of Record has changed, due to the operation of services such as time-of-day routing, presence-based routing or other advanced personalised routing services
- The user may simply have switched to using a different device
- The user's preferences may have changed: the user may have manually changed configuration, or automatic changes may be triggered e.g. by presence or location services (e.g. "disable video when in pub" service).
- Connectivity changes
- Software or hardware updates
- Denied IMS service requests

Once a UE has detected that the cached information is out-of-date then the UE will exchange the capability information in order to refresh its cache.

The problem of out-of-date cached information can be mitigated as follows:

- When communication is established between two UEs, each UE checks its internal cache of the capability information from the other UE. This information includes a record of the point in time at which it was obtained.
- If the UEs own capabilities have changed since that time, then a capability exchange is initiated.

There are three different mechanisms identified that could be used: Presence-based mechanism, SIP OPTIONS based mechanism, and SIP INVITE based solution. Presence is described in 3GPP TS 23.141[z], whilst use of the SIP OPTIONS method and pre-negotiation of services using an INVITE exchange is described by the following sub-clauses. It should be noted that irrespective of the SIP mechanism used for capability exchange, it should be possible to apply operator-desired charging for such exchanges.

An INVITE or OPTIONS request may carry capability information in the sense of caller preferences, as per RFC3841.

Editor's note: It is FFS how caller preferences information would be used to indicate capabilities used by CS/IMS combinational services.

7.2.2.2 OPTIONS

The SIP OPTIONS method allows a UAC to query the capabilities of a UAS. The following guidelines to the handling of SIP OPTIONS aid to limit the number of messages to one exchange (two messages in total) between the two terminals:

- The UAC should send the OPTIONS request with an SDP body describing the terminal capabilities;
- The UAS should copy this information to its cache. The UAS should respond to the request with its terminal capabilities.

A non-CSI client may not expect to find the capabilities of the CSI client in the OPTIONS request and may therefore ignore them, but it should still send a useful response as defined in RFC 3261[x]. If such a client needs to know the capabilities of the CSI client it then sends its own OPTIONS request as defined in RFC 3261[x].

To limit the exchange to two messages between two CSI clients it should be specified that the CS originating UE (A) should start sending the OPTIONS request immediately after the setup of the CS call and the CS terminating UE (B) should not send an OPTIONS request unless it does not receive an OPTIONS request from A within a certain time period, e.g. 10 seconds.

As noted in RFC3261[x] and RFC3264[y], the OPTIONS method is not handled in the same way by proxies as INVITE and it cannot indicate allowed parameter ranges, combinations etc. In particular, when requests are forked for INVITE, multiple early dialogs may result from the return of multiple 183 messages with different SDPs, whereas for OPTIONS, only a single 200 OK will be returned.

7.2.2.3 INVITE

Media components in the SDP of an INVITE may be marked as 'inactive'. This indicates that they are effectively 'on hold' and no media will be sent or received. According to RFC3264, the SDP answer to an 'inactive' media stream must always be marked 'inactive', but is otherwise constructed as if the offer had been 'sendrecv' i.e. the media parameter negotiation takes place as normal giving the endpoints a complete picture of the supported service.

According to normal 3GPP procedures, the SDP in the INVITE will indicate local pre-conditions which prevent the terminating party from being alerted.

Sending an INVITE in this way thus provides a means to determine exactly what services can be supported between two endpoints at a given point in time using four messages (INVITE, 183, PRACK, 200 OK (PRACK)).

If this determination is done whilst a CS call between the endpoints is in progress, then the session may be left in a 'held' state to provide for easy addition of services if requested by the user. Alternatively, the session may be cleared with a CANCEL request to reduce the requirement for the IMS network to store state information.

If this determination is done outside a CS call, then the session may be immediately cleared with a CANCEL request.

7.2.3 Alternative 3

Clause 7.1 describes the type of capability information that is useful to exchange in order to facilitate the introduction of combinational services. Based on the categorization of information in clause 7.1, the following mechanisms are recommended to be introduced for capability exchange:

1. For detecting terminal capabilities and user preferences it is recommended to use a SIP-based mechanism. As per 3GPP TS 23.141[z], Presence provides the required mechanisms to exchange capability information between Presence buddies (i.e. between users that are watching each other).
   Users that don't have Presence enabled for watching each other, or do not get the required capability information via Presence, can use direct SIP-based communications to detect each other's capabilities. Details of such SIP-based mechanisms are described in clause 7.2.2. Note that the capabilities exchanged via such SIP-based mechanisms might be restricted by the possible session policies applied by the IMS operator.

Editor's note: The exact impacts of session policies to SIP based capability exchange is FFS.

2. For detecting the radio network capabilities of the peer user in the scenario that the CS call is created first, it is recommended to study if a mechanism that uses CS call control information element(s) (i.e. elements of TS 24.008 CS call setup) can be developed. One possibility could be to use the Sub-address field or UUS Service 1in the Setup and Connect messages to indicate the, for example, DTM, UTRAN, etc… capabilities of the serving cell. It shall be studied at what stage the capabilities of the serving cell become available to the terminal, i.e. will these capabilities be known before the Setup/Connect message is sent?

3. For detecting the radio network capabilities of the peer user in the scenario that the IMS session is created first, it is recommended to study if a SIP-based mechanism to exchange these capabilities (e.g. P-Access-Network-Info header) can be developed. This header field needs to indicate the radio access of the serving cell, for example DTM, UTRAN, etc.

8 Conclusion and recommendations

Editor's Note: This section will contain the conclusion, if any, of the study

8.1 Conclusion

Four alternatives are described in clause 6 of this TR. The last Alternative D is a converged approach of different aspects that are described in Alternative A, B and C. It is recommended to proceed with the standardization of Alternative D.

A terminal/radio capability exchange mechanism is part of the described alternatives. More details of the different terminal/radio capability mechanisms are described in clause 7.

The principle of CSI phasing is introduced into this TR 23.899 to complement the described phasing in TR 22.979.

Phase 1 will address those capabilities that are already supported by the current set of specifications or require limited additional standardization. The Phase 1 capabilities are Radio Capability Exchange, SIP based capability exchange, adding IMS media component towards an ongoing call and adding CS speech call towards an ongoing IMS session. Phase 1 will focus particularly on end-to-end exchange of terminal capabilities. It also comprises standardisation of exchange of E.164 numbers in SIP, if it can be accomplished in time.

Phase 2 will address those aspects that will require substantial standardization. In Phase 2, additional network functionalities might be required to support standardised procedures for IMS control over CS bearers, including interworking with pure VoIP terminals within or outside the IMS. Phase 2 will use the CSI terminal capabilities standardised in Phase 1. It will be determined in Phase 2 to which extent the additional network functionalities need standardisation in order to avoid options and allow interoperability.

It is recommended to proceed with the standardization of CSI Phase 1 to address the urgent market needs to make IMS services available to the end-users. The minimum aspects required for deployment of CSI Phase 1 shall be addressed.

It is recommended to describe in more detail the additional capabilities required for CSI Phase 2. In this context, further study on the standardization implications of the IMS control over CS bearers and Interworking with legacy, Phase 1 and VoIP terminals will be required.

Editor's Note: The conclusion of Phase 2 is provisional and work on Phase 2 will be continued within this TR

For the initial phase, it is recommended to use the same E.164 number in CS domain and IMS to facilitate the deployment of combinational services. If the same E.164 number is used in IMS and CS domain, it shall be possible to deliver both the IMS session and the call destined to the E.164 number of the UE.

8.2 Standardization Recommendations

When addressing the choice between the SIP INVITE and SIP OPTIONS, the SIP OPTIONS shall be used as the only explicit Terminal Capability Exchange mechanism. This does not preclude the use of Presence if available.

To exchange E.164 numbers during a SIP session establishment, it is recommended to standardize new SDP "a=" attribute values or have the current "c=" line extended by new network type and address type. The SDP extensions shall be for widespread use and therefore it is also recommended that the IETF MMUSIC will specify an appropriate RFC.

Annex A: Possible CSI phased approach

The following table indicates the impact on standards, terminals and networks of each of the proposals in Section 6.3 (Alternative C) and Section 6.4 (Alternative D).

It is proposed that standardization of CSI in a first phase is focused on the issues with limited standardization and implementation impact. The other issues could be considered in a second phase.

Subclause and Issue	**3GPP Standardisation Impact**	**Other Standardisation impact**	**UE impact**	**Core network impact**	**Proposed Conclusion**
Alternative D 6.4.1 Radio Capability Exchange	Define mechanism for exchange of radio capabilities in SA2 and format in CN WGs	None	Yes	No	Phase 1 if feasible
Alternative D 6.4.1 SIP based capability exchange	For OPTIONS, describe use of SDP For INVITE, describe use of inactive	For OPTIONS, verify that this is RFC 3261/3264 compliant (so that it can be used in interworking scenarios)	Yes	No for OPTIONS Yes for INVITE, optimisation of implementation for a large number of inactive INVITEs	Phase 1
Alternative D 6.4.2 capability exchange in INVITE	Describe use of inactive for supported but not requested capabilities Open issue: radio capabilities within the SIP/SDP?	IETF: define E.164 number communications in SIP/SDP (unless done in 3GPP) Open issue: radio capabilities within the SIP/SDP?	Yes	optimisation of implementation for a large number of inactive INVITE components	E.164 indication not in Phase 1
Alternative D 6.4.3 Addition of IMS media component	None	None	Only combination with CS call	None	Phase 1
Alternative D 6.4.4 Addition of CS call	None on top of 6.3.2	None on top of 6.3.2	Only combination with IMS session	None	Phase 1 assuming MS-ISDN is available
Alternative D	Describe CBCF	6.3.2 + Indication	Yes (6.3.2/4	Yes (CBCF)	Not Phase 1

6.4.5 and 6.4.6	function? Indication for CS call direction in cases 6.3.5.2 and 6.3.6.1 Consideration of alternative approaches	for CS call direction?	+ indication for CS call direction)		
Alternative D 6.4.7	(tbd)	(tbd)	(tbd)	(tbd)	To be provided for each phase
Alternative C	Describe Roles of AS/IWF	IETF: define E.164 number communications in SIP/SDP (unless done in 3GPP)	Yes, similar to Alternative D, 6.4.5 and 6.4.6	Yes (new AS, IWF)	Not Phase 1

It looks like the approaches in Alternative D (6.4.5 and 6.4.6) and Alternative C could be built upon the same standardised indication of an E.164 number in the SDP in a second CSI phase.

Annex B: CSI use cases

The following table depicts the use case considered by each of the proposals in Section 6. It also shows how each subsection for alternative D is related to the previous sections for alternative A, B and C.

	Network control	**Client control**
Between CSB UEs	Section 6.3 (alternative C) **Section 6.4.7 (alternative D)**	Section 6.1.3.4 (alternative A e2e case) Section 6.2 (alternative B) **Section 6.4.3 (alternative D)** **Section 6.4.4 (alternative D)**
Between a CSB UE and a pure IMS VoIP UE	Section 6.1.3.2 (alternative A e2g network control case) **Section 6.4.5 (alternative D)** **Section 6.4.6 (alternative D)**	Section 6.1.3.3 (alternative A e2g client control case) **Section 6.4.8 (alternative D)**

Table B-1: Use cases of alternatives for CSI

Annex C: Change history

Editor's note: pre-approval change history to be removed when/if TR is approved.

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
14/4/2004					*First draft*	---	*0.0.0*

28/4/2004					*TR updated to include changes agreed in S2-041591 at SA2#39*	*0.0.0*	*0.1.0*
26/5/2004					*TR updated to include changes agreed at SA2#40 in S2-042214,S2-042217, S2-042238, S2-042239 and S2-042240.*	*0.1.0*	*0.2.0*
06/9/2004					*TR updated to include changes agreed at SA2#41 in S2-042723, S2-04296, S2-042868. S2-042869, S2-042870, S2-022907, S2-042882, S2-042908*	*0.2.0*	*0.3.0*
25/10/2004					*Tr updated to include changes agreed at SA2#42 in S2-043257, S2-043258, S2-043347, S2-04334, S2-043371, S4-043372, S2-043373, S2-043350, , S2-043381*	*0.3.0*	*0.4.0*
1/11/2004					*Editorial updates*	*0.4.0*	*0.4.1*
25/11/2004					*TR updated to include changes agreed at SA2#43 in S2-043555, S2-043783, S2-043785, S2-043786, S2-043825, S2-043826, S2-043889, S2-043787*	*0.4.1*	*0.5.0*
25/02/2005					*TR updated to include changes agreed at SA2#44 in S2-050421, S2-050442, S2-050462, S2-050513*	*1.0.0*	*1.1.0*
21/06/2005					TR updated to include changes agreed at SA2#46 in S2-051283, S2-051286, S2-051330, S2-051348	1.1.0	1.2.0