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SERIES Q: SWITCHING AND SIGNALLING

Signalling requirements and protocols for the NGN – Service and session control protocols – supplementary services

Signalling requirements and protocol profiles for NGN customized ringing tone service

Recommendation ITU-T Q.3611

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Recommendation ITU-T Q.3611

Signalling requirements and protocol profiles for NGN customized ringing tone service

Summary

Recommendation ITU-T Q.3611 specifies the signalling requirements and protocol profiles for next generation network (NGN) customized ringing tone (CRT).

Source

Recommendation ITU-T Q.3611 was approved on 29 June 2009 by ITU-T Study Group 11 (2009-2012) under Recommendation ITU-T A.8 procedures.

Keywords

Call server, CRT, IMS, NGN, SDP, SIP.

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1 Scope

This Recommendation specifies signalling requirements and protocol profiles for NGN customized ringing tone (CRT) service. This Recommendation also contains signalling flows.

The CRT service is defined in the next generation network (NGN), including IP multimedia subsystem (IMS) and call-server based networks. This Recommendation is based on [b-ITU-T Y.2214] and covers in this version only the signalling requirements and protocol profiles for the presentation features of CRT. The capability for the CRT filtering features is for further study.

This Recommendation covers the protocol profiles over user network interface (UNI) and network-to-network interface (NNI) based on [ITU-T Q.3402] and [ITU-T Q.3401], respectively. Detailed protocol profiles for other interfaces may be subject to future specifications.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T Q.763] Recommendation ITU-T Q.763 (1999), *Signalling System No. 7 –ISDN User Part formats and codes*.

[ITU-T Q.3401] Recommendation ITU-T Q.3401 (2007), *NGN NNI signalling profile (Protocol Set 1)*; Amendment 1 (2008).

[ITU-T Q.3402] Recommendation ITU-T Q.3402 (2008), *NGN UNI signalling profile (Protocol Set 1)*.

[ITU-T T.81] Recommendation ITU-T T.81 (1992) | ISO/IEC 10918-1:1994, *Information technology – Digital compression and coding of continuous-tone still images – Requirements and guidelines*.

[ITU-T Y.2012] Recommendation ITU-T Y.2012 (2006), *Functional requirements and architecture of the NGN release 1*.

[ITU-T Y.2701] Recommendation ITU-T Y.2701 (2007), *Security requirements for NGN release 1*.

[IETF RFC 2616] IETF RFC 2616 (1999), *Hypertext Transfer Protocol – HTTP/1.1*.

[IETF RFC 3261] IETF RFC 3261 (2002), *SIP: Session Initiation Protocol*.

[IETF RFC 3262] IETF RFC 3262 (2002), *Reliability of Provisional Responses in the Session Initiation Protocol (SIP)*.

[IETF RFC 3311] IETF RFC 3311 (2002), *The Session Initiation Protocol (SIP) UPDATE Method*.

[IETF RFC 3372]	IETF RFC 3372 (2002), *Session Initiation Protocol for Telephones (SIP-T), Context and Architectures*.
[IETF RFC 3515]	IETF RFC 3515 (2003), *The Session Initiation Protocol (SIP) Refer Method*.
[IETF RFC 3959]	IETF RFC 3959 (2004), *The Early Session Disposition Type for the Session Initiation Protocol (SIP)*.
[IETF RFC 3960]	IETF RFC 3960 (2004), *Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)*.
[IETF RFC 5009]	IETF RFC 5009 (2007), *Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media*.

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following term defined elsewhere:

3.1.1 early media [IETF RFC 3959]: The media (e.g., audio and video) that is exchanged before a particular session is accepted by the called user.

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 called CRT service subscriber: The called party who subscribes to the called party configured CRT service.

3.2.2 calling CRT service subscriber: The calling party who subscribes to the CRT service.

3.2.3 CRT service user: The called party who experiences the CRT service.

3.2.4 early media for CRT: Early media that are delivered to the called party with the media contents configured by the service subscriber. It can be generated by the originating network or the terminating network, depending on the location of service subscriptions.

3.1.5 icon ring tone server (IRT server): Icon ring tone server (IRT) stores the ringing tone media such as audio, video and picture files. The IRT server behaves as a web server and can be provided by content providers or operators. A user terminal can interact with the IRT server via, for example, HTTP, to download the ringing tone media.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

AS	Application Server
AS-FE	Application Support Functional Entity
AVP	Audio Video Profile
CDIV	Communication DIVersion
CFNR	Call Forwarding on No Reply
CRT	Customized Ringing Tone
CSN	Circuit Switched Network
ECT	Explicit Communication Transfer
HOLD	Communication HOLD

HTTP	HyperText Transfer Protocol
IAM	Initial Address Message
IBC-FE	Interconnection Border Gateway Control Functional Entity
I-CSC-FE	Interrogating Call Session Control Functional Entity
IMS	IP Multimedia Subsystem
IRT	Icon Ring Tone
MGC-FE	Media Gateway Control Functional Entity
MRC-FE	Media Resource Control Functional Entity
MRFC	Media Resource Function Controller
MRP-FE	Media Resource Processing Functional Entity
NGN	Next Generation Network
NNI	Network-to-Network Interface
OIP	Originating Identification Presentation
OIR	Originating Identification Restriction
P-CSC-FE	Proxy Call Session Control Functional Entity
RTCP	RTP Control Protocol
RTP	Real-Time Transport Protocol
S-CSC-FE	Serving Call Session Control Functional Entity
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP UA	SIP User Agent
SUP-FE	Service User Profile Functional Entity
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UE	User Equipment
UE-A	User Equipment for user A
UE-B	User Equipment for user B
UNI	User-to-Network Interface
URL	Uniform Resource Locator
UUI	User-to-User Information

5 CRT service and architecture

5.1 General description

The customized ringing tone (CRT) service provides the called party with the calling party's customized ringing tone. The service can be subscribed by the calling party or the called party. The service is invoked to provide customized ringing tone to the called party, according to the service configuration. The CRT contents can be multimedia representing avatar, business card, etc. The calling party subscribed to CRT is always triggered at the originating network's service system. Otherwise, the called party subscribed to CRT is always triggered at the terminating network's service system.

To protect the called party from possible malicious CRT contents, the network should provide the capability for the called party to reject CRT (e.g., reject CRT unconditionally, for unknown parties, for black listed parties, etc.). The capability for the CRT filtering features is for further study.

5.2 General architecture

The CRT service is defined and provided in the NGN architecture, which is either IMS-based or call-server based. Figure 5-1 shows a general architecture that is compliant with [ITU-T Y.2012] for providing the CRT service based on IMS and call-server networks, respectively.

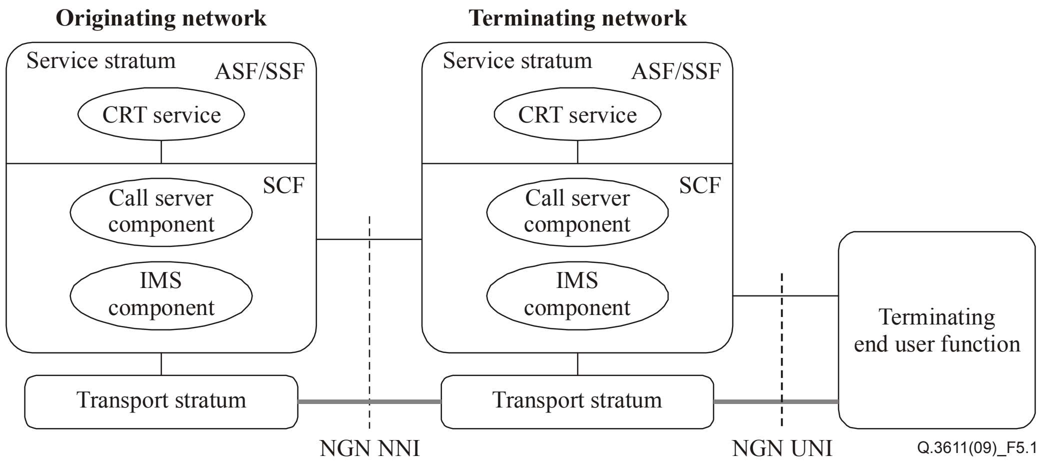

Figure 5-1 – CRT service functional architecture

5.3 Possible CRT service architectures

Originating IMS-based and call-server based possible service architectures for the CRT service are shown in Figures 5-2 and 5-3, respectively. The terminating IMS-based possible scenarios for the calling party configured CRT service are shown in Figure 5-4.

These figures show general scenarios assuming that the originating network is different from the terminating network. Although the functional architectures are different for IMS- and call-server based networks, the service control mechanisms are identical.

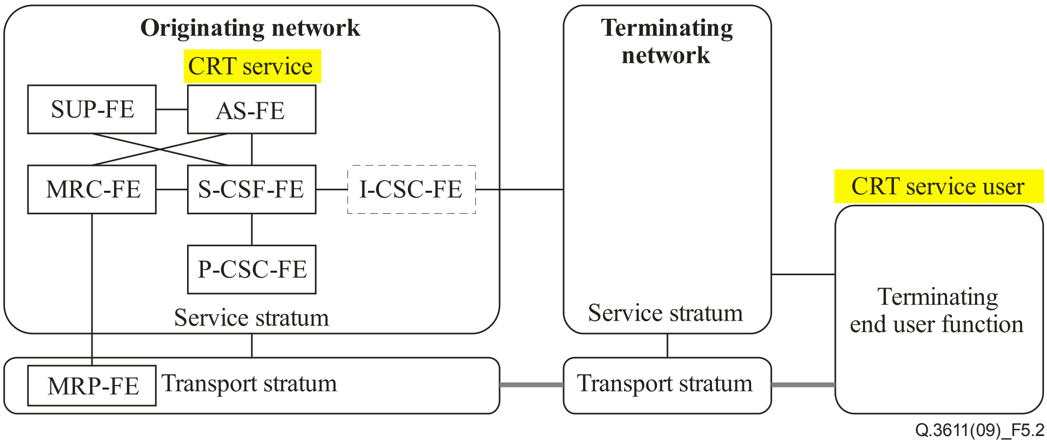

Figure 5-2 – Originating IMS-based CRT service architecture

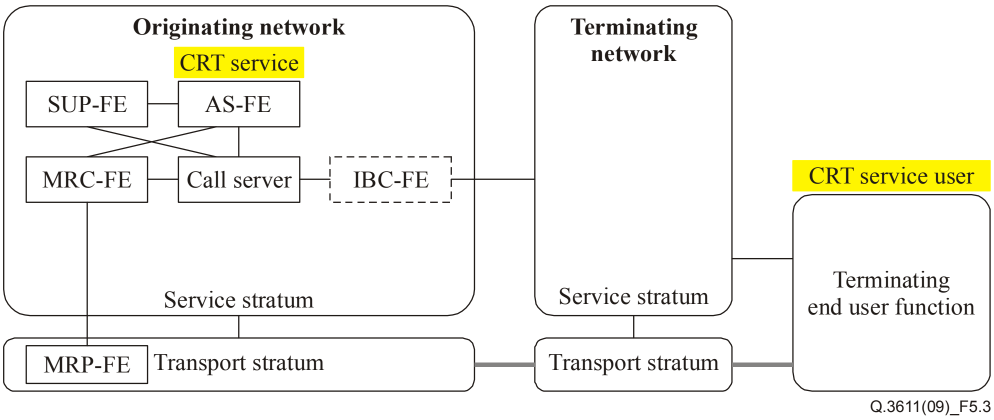

Figure 5-3 – Originating call-server based CRT service architecture

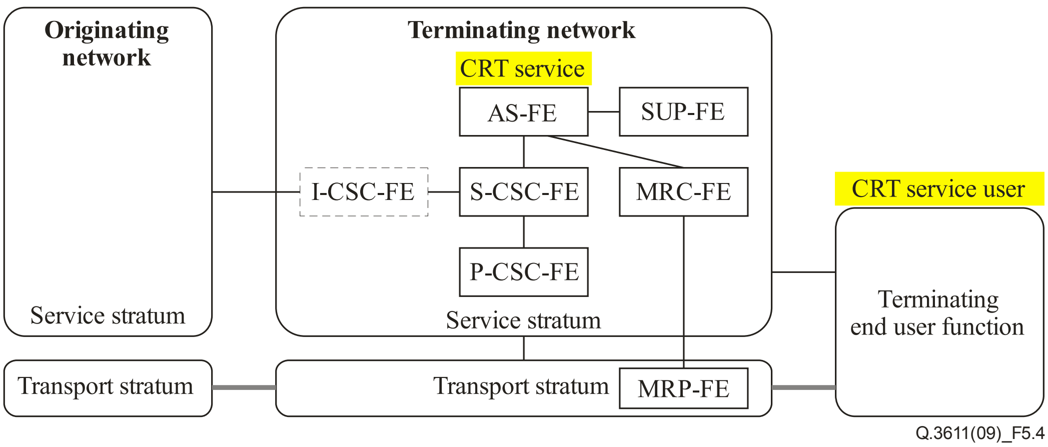

Figure 5-4 – Terminating IMS-based CRT service architecture

6 Service models

There are three models for ringing tone generations. The CRT service is provided based on at least one of these service models, and can be provided with more than one model.

Gateway model is to establish early media sessions in the same way as regular sessions, using an offer/answer exchange. The gateway model consists of managing early media sessions using offer/answer exchanges in reliable provisional exchanges, PRACKs, and

	UPDATEs [IETF RFC 3960]. In case of non-reliable provisional exchanges, re-INVITE can be used.
Application server model	is to establish early media by distinguishing early sessions from regular sessions by using the early-session option tag. By indicating support of the early-session disposition type in the user agent server (UAS), the user agent client (UAC) knows that offer/answer exchanges for early media (early-session disposition type) are kept separate from regular media (session disposition type) [IETF RFC 3960].
HTTP model	is to provide the web URL of the subscriber's repository for CRT contents to be presented during early session. This information is filled by and delivered from the CRT service system, within the initial INVITE.

7 Media formats

In order to guarantee a certain level of early media support and compatibility between the calling and called party terminals, appropriate media formats shall be negotiated or selected to play the CRT contents.

7.1 Audio

If audio is supported, the codec list shall be as specified in clause 8.1 of [ITU-T Q.3401] and [ITU-T Q.3402].

7.2 Video

If video is supported, the codec list shall be as specified in clause 8.1 of [ITU-T Q.3401] and [ITU-T Q.3402].

7.3 Text

If plain text is supported, any character encoding (charset) that contains a subset of the logical characters (e.g., Unicode [b-Unicode]) may be used.

7.4 Still image

If still image is supported, JPEG encoding in compliance with [ITU-T T.81] shall be supported. In addition, the interchange format described in [b-JFIF] may be used with JPEG. Other encoding formats (e.g., GIF ([b-GIF87a], and [b-GIF89a]), or PNG ([b-IETF RFC 2083])) may be supported.

8 Signalling requirements

The following clauses describe the signalling requirements for providing CRT services in each functional entity.

8.1 Requirements for the terminating UE

8.1.1 Requirements for supporting CRT on gateway model

If the terminating user equipment (UE) receives an initial INVITE request, and wishes to receive early media, it should:

a) Send out SDP answer via a response bearing a 18x code to the initial INVITE request towards the originating side, or

b) Send SDP answer in the response of UPDATE [IETF RFC 3311] method.

8.1.2 Requirements for supporting CRT on application server model

If the terminating UE wishes to receive early media authorization descriptions, as described in [IETF RFC 3959], when it receives the initial INVITE request which has set the "early-session" option tag in the Require header field, it shall reply with a normal provisional response bearing a 18x code indicating that the terminating UE supports early-session disposition type SDP and should set the "100rel" option tag in the Require header field.

8.1.3 Requirements for supporting CRT on HTTP model

The terminating UE shall support Alert-Info and/or Call-Info header field as specified by [IETF RFC 3261] and [IETF RFC 3960].

If the media file indicated in the Alert-Info or Call-Info header field has been downloaded, the terminating UE should play the downloaded information to override the locally generated ring information, except if the user denies or stops it.

8.1.4 Requirements for supporting multiple CRT service models

When multiple models are applied, the terminating UE may be configured to determine whether to play downloaded information only and/or early media information if different media types are involved, and if the same media type is involved, then the preference is set by the local policy.

8.1.5 Requirements when user answers the communication

For the gateway model, the terminating UE does not need to distinguish between early media and regular media before a final answer is received, so there is no requirement for the terminating UE in this case.

For the application server model, if the terminating UE uses a different media port and/or a connect address for regular media and early media, when the user answers the dialog, the UE should stop playing early media information.

If the HTTP model is applied, the terminating UE should have the local policy to determine whether to keep playing the CRT contents while conversing with the originating UE.

8.2 Requirements for the terminating P-CSC-FE

8.2.1 Requirements for supporting CRT on gateway model

The P-CSC-FE may add, remove, or modify the P-Early-Media header field within the forwarded SIP requests and responses, according to the procedures in [IETF RFC 5009].

8.2.2 Requirements for supporting CRT on application server model

The P-CSC-FE may add, remove, or modify, the early-session SDP within the forwarded SIP requests and responses, according to procedures in [IETF RFC 3959].

8.2.3 Requirements for supporting CRT on HTTP model

The P-CSC-FE may have local policy to remove, or modify the Alert-Info, and/or Call-Info header field.

8.3 Requirements for the I-CSC-FE

CRT has no impact on this FE.

8.4 Requirements for the S-CSC-FE

CRT has no impact on this FE.

8.5 Requirements for the IBC-FE

8.5.1 Requirements for supporting CRT on gateway model

The IBC-FE may add, remove, or modify the P-Early-Media header field within the forwarded SIP requests and responses, according to the operator's policy.

8.5.2 Requirements for supporting CRT on application server model

The IBC-FE may add, remove, or modify the early-session SDP within the forwarded SIP requests and responses, according to the operator's policy.

8.5.3 Requirements for supporting CRT on HTTP model

The IBC-FE may have the local policy to remove, or modify the Alert-Info, and/or Call-Info header field.

8.6 Requirements for the call server

CRT has no impact on this FE.

8.7 Requirements for the AS-FE

The requirements in this clause are for invocation and operation of the CRT service. The activation, deactivation, and configuration of the CRT service are out of scope of this Recommendation.

The AS-FE provides CRT service regardless of whether the session systems are call-server or IMS-based.

8.7.1 Requirements for providing CRT service by gateway model

The AS-FE shall apply CRT service after receiving a response bearing a 18x code with SDP answer to the initial INVITE request.

If the CRT service can be applied, the AS-FE shall:

a) Send a CRT SDP offer via PRACK/UPDATE request towards the terminating side; and
b) If the initial INVITE request carries a SDP offer with precondition, send the SDP answer in the response bearing a 18x code towards the originating side.

The AS-FE may not forward the originating side SDP offer in the PRACK/UPDATE request towards the terminating side.

The AS-FE may set all the media types bandwidth in the SDP towards the originating side to 0, in order to avoid RTCP error.

8.7.2 Requirements for providing CRT service by application server model

If the initial INVITE request is received, then the AS-FE shall:

a) If the provisional response has no "early-session" option tag in the Require header field, set the "early-session" option tag in the Require header field;
b) If the provisional response has no "100 rel" option tag in the Require header field, set the "100rel" option tag in the Require header field;
c) Send the INVITE request towards the terminating UE.

If the INVITE request's 420 error response with "early-session" option tag in the Unsupported header field is received, it indicates that the terminating UE does not support early-session, then the AS-FE shall:

a) Provide CRT service by gateway model or HTTP model or multi-dialogue model;

b) Remove the "early-session" option tag in the Require header field in the previous INVITE request;

c) Send the revised INVITE request towards the terminating UE.

If the INVITE request provisional response bearing a 18x code is received, it indicates that the terminating UE supports early-session, then the AS-FE shall:

a) If the provisional response has no "100rel" option tag in the Require header field, set the "100rel" option tag in the Require header field;

b) Send the provisional response towards the originating UE.

If a PRACK request to the INVITE request provisional response bearing a 18x code is received, according to [IETF RFC 3959], the AS-FE shall:

a) If terminating side does not indicate "100rel" in the Require header field, transfer the PRACK request to the UPDATE request;

b) Acquire SDP for CRT from MRFC;

NOTE – The SDP for CRT can be acquired by sending an INVITE request without SDP to MRFC.

c) Add new message body that has early-session disposition type in the PRACK/UPDATE request to contain the acquired SDP;

d) Set "early-session" option tag in the Require header field if it does not exist;

e) Send the PRACK/UPDATE request towards the terminating UE.

If a final answer is received by the AS-FE on a dialog, the AS-FE should stop the early media CRT information on the answering dialog.

8.7.3 Requirements for providing CRT service by HTTP model

If an initial INVITE request is received, then, according to [IETF RFC 3261] and [IETF RFC 3960], the CRT AS-FE is required to act as follows:

a) If an iconic photo, photo explanation text, and/or card information need to be presented on the terminating UE's screen, the AS-FE should add a Call-Info header field in the INVITE request, and, according to [IETF RFC 3261], set the HTTP protocol address of the iconic photo file in the Call-Info header field using "icon" value for the purpose parameter, set the photo explanation text in the Call-Info header field using "info" value for the purpose parameter, and/or set the card information in the Call-Info header field using "card" value for the purpose parameter;

b) If a multimedia information needs to be played on the terminating UE, the AS-FE may add an Alert-Info and/or Call-Info header field, and set the HTTP URL of the multimedia file to the value of the Alert-Info/Call-Info header field;

c) Shall send the INVITE request towards the terminating UE.

NOTE – It is recommended that the size of the multimedia file and photo file be small in order not to delay communication establishment.

8.7.4 Requirements for supporting multiple models

If either gateway model or application server model is applied, the AS-FE should have the policy to determine whether to play downloaded information and/or early media information if different media types are involved, and if the same media type is involved, then preference is set by the policy.

This policy should be configured by the calling CRT service subscriber, the called CRT service subscriber, the originating network's operator or the terminating network's operator.

8.7.5 Requirements when user answers the communication

If the CRT service is provided by the application server model/gateway model, the AS-FE should stop the early media CRT information on the answering dialog.

If the CRT service is provided by the gateway model, before forwarding the final answer, the AS-FE shall:

a) send the originator's SDP offer via a re-INVITE request towards the terminating side, then after getting the SDP answer from the terminating side, send it via a "200 OK" response towards the originating side; or

b) query the terminator's SDP offer by sending a re-INVITE request without SDP towards the terminating side, then after getting the SDP offer, send it via a "200 OK" response towards the originating side.

If the CRT service is provided by the HTTP model, there is no requirement.

9 Protocol profiles

The protocol profiles for CRT service shall be based on the following ITU-T Recommendations.

Table 9-1 – Base Recommendations for CRT protocol profiles

Profile	Interval	Protocol	Recommendation
NGN UNI	Between the terminating end user function and the service control function in the terminating network	SIP/SDP	[ITU-T Q.3402]
NGN NNI	Between the service control functions of originating and terminating networks	SIP/SDP	[ITU-T Q.3401]

This clause provides CRT service specific messages and headers among the above Recommendations and relevant RFCs.

Other messages and headers that are not specific for CRT may be supported, as specified in the base Recommendations in Table 9-1, if they are not explicitly specified in this Recommendation.

The three service models (gateway, application server and HTTP) defined in clause 6 are not applicable to legacy terminals (e.g., terminals which are not attached via NGN UNI) in the terminating network.

9.1 IMS-based CRT service

9.1.1 Protocol profiles for gateway model

If early media is established with the gateway model, the UPDATE method shall be used as defined in [IETF RFC 3311].

If early media connection is established by the exchanges of initial INVITE and non-reliable provisional response, after finishing early media session, regular session can be made by media re-negotiation exchanging second Offer SDP and Answer SDP after completion of the first dialogue.

Unidirectional early media may also be supported by responding to initial INVITE with non-reliable provisional response before sending actual media. The regular session is then established by responding with final SDP answer to the initial INVITE.

Table 9-2 illustrates the SIP messages that are specific to the CRT service. Other SIP messages may be supported as specified in the base Recommendations in Table 9-1.

Table 9-2 – CRT service specific methods and headers for gateway model

a) Early session establishment

Method/response	Code	Header	Code	Option tag	References
UPDATE	M	–	–	–	RFC 3311

b) Regular session establishment

Method/response	Code	Header	Code	Tag	Reference
UPDATE – 200 OK	M	–	–	–	RFC 3311
re-INVITE – 200 OK	O	–	–	–	RFC 3261

In order to make regular dialogue after early dialogue finishes, there are two possibilities:

a) final response 200 OK delivery regular Answer

b) re-establishment of session with re-INVITE

See clause I.1 for detailed signalling flows and messages.

9.1.2 Protocol profiles for application server model

9.1.2.1 SIP profile

Initial INVITE shall include SDP. If early media session is established with the reliable provisional response, the PRACK method shall be used as defined in [IETF RFC 3262].

If early media session is provided, the option tag "early-session" shall be supported, the disposition type "early-session" shall be supported, and the content type "multipart/mixed" may be supported to specify distinctive session types (e.g., regular session and early session) within a single message, as defined in [IETF RFC 3959].

Table 9-3 illustrates the SIP messages that are specific to the CRT service. Other SIP messages may be supported as specified in the base Recommendations in Table 9-1.

Table 9-3 – CRT service specific methods and headers for application server model

SCF/CRT AS-FE → Terminating EUF					
Request method	**code**	**Header**	**code**	**Description**	**Reference**
INVITE	M	Supported	M	Supported: 100rel, early-session	RFC 3262
		Require	M	Require: 100rel, early-session	
		Content-Type	O	Content-Type: multipart/mixed	
PRACK	M	RAck	M	–	

Terminating EUF → SCF/CRT AS-FE					
Response method	**code**	**Header**	**code**	**Description**	**Reference**
18x	M	RSeq	O	–	RFC 3262 RFC 3959
		Require	O	Require: 100rel, early-session	

9.1.2.2 SDP profile

Different types of media such as audio, video, text and image can be utilized as different ringing tones. Each media type shall be specified in "m=" line (i.e., m=text, m=audio, m=video).

9.1.3 Protocol profiles for HTTP model

If the CRT URL information is delivered in provisional response, INVITE shall include Call-Info header and 'purpose' parameters or/and Alert-Info header as defined in [IETF RFC 3261].

Table 9-4 illustrates the SIP messages that are specific to the CRT service

Table 9-4 – CRT service specific methods and headers for HTTP model

SCF/CRT AS-FE → Terminating EUF					
Method/response	**Code**	**Header**	**Code**	**Option tag**	**Reference**
INVITE	M	Call-Info	M	purpose	RFC 3261
		Alert-Info	O	–	

The called party shall support HTTP/1.1 as defined in [IETF RFC 2616] in order to get CRT contents.

See clause I.3 for signalling flows and message usages.

9.2 Call-server based CRT service

For the originating call-server based network, the protocol profiles described in clause 9.1 are applicable for the service models : gateway model, application server model, and HTTP model.

For the terminating call-server based network, CRT is not applicable.

10 Interactions between networks

10.1 Providing circuit switched network CRT to the terminating UE in the NGN domain

For the CRT service activated in circuit switched networks (CSN), upon receipt of the initial address message (IAM) with the CRT text in UUI [ITU-T Q.763], and if the MGC-FE supports the Call-Info header as a UUI mapping option, and has received the CRT text in the IAM request, then the MGC-FE shall send INVITE with CRT content information in Call-Info header.

Service providers may send a service indicator in UUI to MGC-FE, and MGC-FE may retrieve the CRT contents.

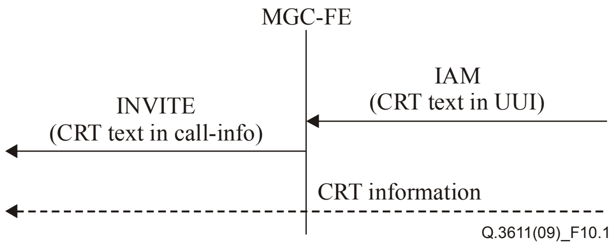

Figure 10-1 – Signalling interaction from CSN to NGN

10.2 Providing NGN CRT to the called party in the circuit switched network domain

For the CRT service activated in the NGN, upon receipt of the INVITE with the CRT text in Call-Info header, and if the MGC-FE supports the UUI as a network option, and has received the CRT text in the INVITE request, then the MGC-FE shall send IAM with a CRT text in UUI.

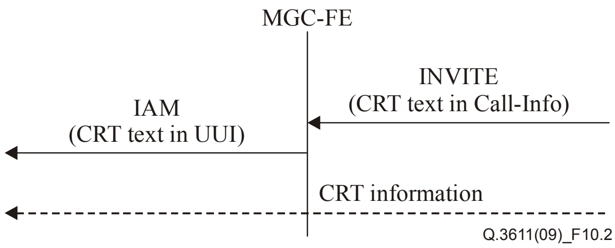

Figure 10-2 – Signalling interaction from NGN to CSN

Service providers may send a service indicator in the Call-Info header to MGC-FE, and MGC-FE may retrieve the CRT contents.

11 Interactions with other services on NGN

11.1 Originating identification presentation (OIP)

No impact on CRT service.

11.2 Originating identification restriction (OIR)

If the CRT service is associated with the originator's id information, CRT AS-FE shall not apply the service on the communication if OIR service has restriction rules on the originating identity information.

11.3 Terminating identification presentation (TIP)

No impact on CRT service.

11.4 Terminating identification restriction (TIR)

No impact on CRT service.

11.5 Communication HOLD (HOLD)

No impact on CRT service.

11.6 Call Waiting (CW) services

In case of B having an existing session with C, and A calling B with CRT service, B experiences either the CRT service or the default communication waiting indication according to the operator's policy or subscription options.

11.7 Communication diversion (CDIV) services

11.7.1 General

No impact on CRT service.

If the diverting party has CDIV active, and the calling party has CRT service active, then the CRT service of the calling party shall be applied to the session by the AS-FE providing the CRT service of diverted-to party, except in the case of call forwarding on no reply (CFNR).

If the diverting party has CDIV active, and the calling party has CRT service active, then the CRT of the calling party should be applied to the diverted-to user, if the diverted-to user has subscribed to the CRT service to receive such services.

11.7.2 Call forwarding on no reply (CFNR)

If the diverting party has CFNR active, and the calling party has CRT service active, then the CRT service of the calling party shall be applied to the session by the AS-FE providing CRT service for the calling party until the CFNR timer expires. Upon the CFNR timer expiry, the AS-FE providing the CRT service for the calling party shall deliver the CRT service to the diverted-to party.

11.8 Explicit communication transfer (ECT) service

If the calling party has CRT service active, the AS-FE providing CRT service shall apply the calling party's CRT to the transfer target in the case of blind transfer. The AS-FE providing the CRT service shall not apply the calling party's CRT in the case of consultative transfer while the call is being transferred to the transfer target.

11.9 Originating CRT service interacting with terminating CRT service

If the originating CRT service is triggered and the terminating CRT service is also triggered, and as both services provide early media to the called party, a conflict occurs.

In this case, some priority mechanism is required and the priority mechanism within SIP is for further study.

Another mechanism is to configure the priority between service operators by pre-arrangement. The configuration is set on the service profiles by subscribers or operators. In this case, since the called party is usually the subscriber of the terminating CRT, the terminating CRT has higher priority over the originating CRT.

The signalling flows for priority service between the originating and terminating CRT is presented in clause I.4.

12 Security considerations

The CRT service is required to use appropriate security mechanisms to meet the general security requirements of NGN [ITU-T Y.2701]).

Also, the CRT service provides early/regular media transport and signalling messages, the NGN network infrastructure for the CRT service should ensure the confidentiality and integrity of the signalling flows transported on it.

Therefore, it is recommended to provide the transport and/or network security (respectively, e.g., TLS [b-IETF RFC 2246] and/or IPSec [b-IETF RFC 2401]) and/or the application security (such as

S/MIME [b-IETF RFC 2633]) for signalling messages using SIP between two endpoints, as described in the security considerations of the core SIP specification [IETF RFC 3261], the Early Media and Ringing Tone Generation specification [IETF RFC 3960], and Session Initiation Protocol for Telephones (SIP-T [IETF RFC 3372]).

In the gateway model and the application server model, the transport and network security and/or the application security should be applied.

In the HTTP model, the application-layer security should be applied, because the body of the signalling message (e.g., Alert-info of SIP [IETF RFC 3261]) contains the web URL. Placing the URL in the header field can pose a security risk. If a called party (UE-B) fetches the URLs provided by a malicious caller (UE-A), the called party may be at risk for displaying inappropriate or offensive content, dangerous or illegal content, and so on. Hence, the application security (e.g., S/MIME) should be applied in order to provide the message authentication mechanism.

Annex A

Multi-dialog mode

(This annex forms an integral part of this Recommendation)

A.1 General description

Multi-dialog model establishes both early and regular dialog together during the establishment of the communication. Since it uses a forking mechanism for making both dialogs, they are correlated with the CRT service.

A.2 Signalling requirements

A.2.1 Requirements for the terminating UE

If the terminating UE supports the P-Early-Media header, and wishes to receive early media authorization indications as described in [IETF RFC 5009], upon receiving a REFER request on the early dialog created by the initial INVITE request, whose Refer-to header field has P-Early-Media header parameter with the value of "supported", the terminating UE shall:

a) Apply normal REFER handling procedures according to [IETF RFC 3515];

b) Add a P-Early-Media header field with the value of "supported" in the resulting INVITE request;

c) If the user limits the early media type, add a session SDP to the resulting INVITE request, which includes all the media types that the user permits; otherwise, no session SDP is added to the INVITE request.

If the user answers a dialog, the terminating UE may send a CANCEL request to terminate the dialog(s) that result from the REFER request on the early dialog, whose Refer-to header field has P-Early-Media header parameter.

A.2.2 Requirements for the AS-FE

If the INVITE request provisional response bearing a 18x code without P-Early-Media header and without "early-session" option tag in the Supported header is received, then the AS-FE shall:

a) Create a REFER request on the dialog created by the provisional response;

b) Create a CRT Session Identifier URI addressed to this AS-FE. The URI shall be created in such a way that a new dialog set-up towards this URI can be easily correlated with the current REFER dialog;

c) Set the Refer-to header field of the REFER request to the value of the CRT Session Identifier URI, and add a P-Early-Media header parameter with the value of "supported"; (This will help the CRT AS-FE to terminate the early dialog created by the result INVITE request.)

d) Send the REFER request towards the terminating UE.

NOTE 1 – It depends on the operator's policy or user configuration whether the CRT information needs to be provided to all the terminators, or some terminators, or only one terminator, if the INVITE request has been forked.

If a new initial INVITE request is received targeting the CRT Session Identifier URI created earlier, according to [IETF RFC 5009], the AS-FE shall:

a) Acquire SDP for CRT from MRFC;

NOTE 2 – The SDP for CRT can be acquired by sending an INVITE request without SDP to MRFC.

b) Create a provisional response bearing a 18x code to the INVITE request;

c) Add a P-Early-Media header field in the created provisional response, the value of the P-Early-Media header field has one of the following values: "sendonly", or "inactive";

d) Add a message body that has session disposition type to the created provisional response, if there is no session SDP in the INVITE request, the body will contain the acquired SDP for CRT; otherwise, based on the acquired SDP for CRT, the body will contain the SDP answer to the SDP offer of the INVITE request;

e) Send the created provisional response towards the terminating UE.

If a final answer is received on a dialog, the AS-FE should stop the early media CRT information on the dialog between the terminating UE and a CRT Session Identifier URI, which is correlated to the answering dialog, or, may terminate the correlated dialog by sending a 486 (Busy) response.

Appendix I

Signalling flows

(This appendix does not form an integral part of this Recommendation)

The detailed message examples are not shown for simplicity and most of the headers are shown in the first message which includes the significant ones.

I.1 CRT service using gateway model

Figure I.1 signalling flow shows an IMS-based CRT service provided by the gateway model scenario:

NOTE – This signalling flow shows the case of regular session establishment with re-INVITE.

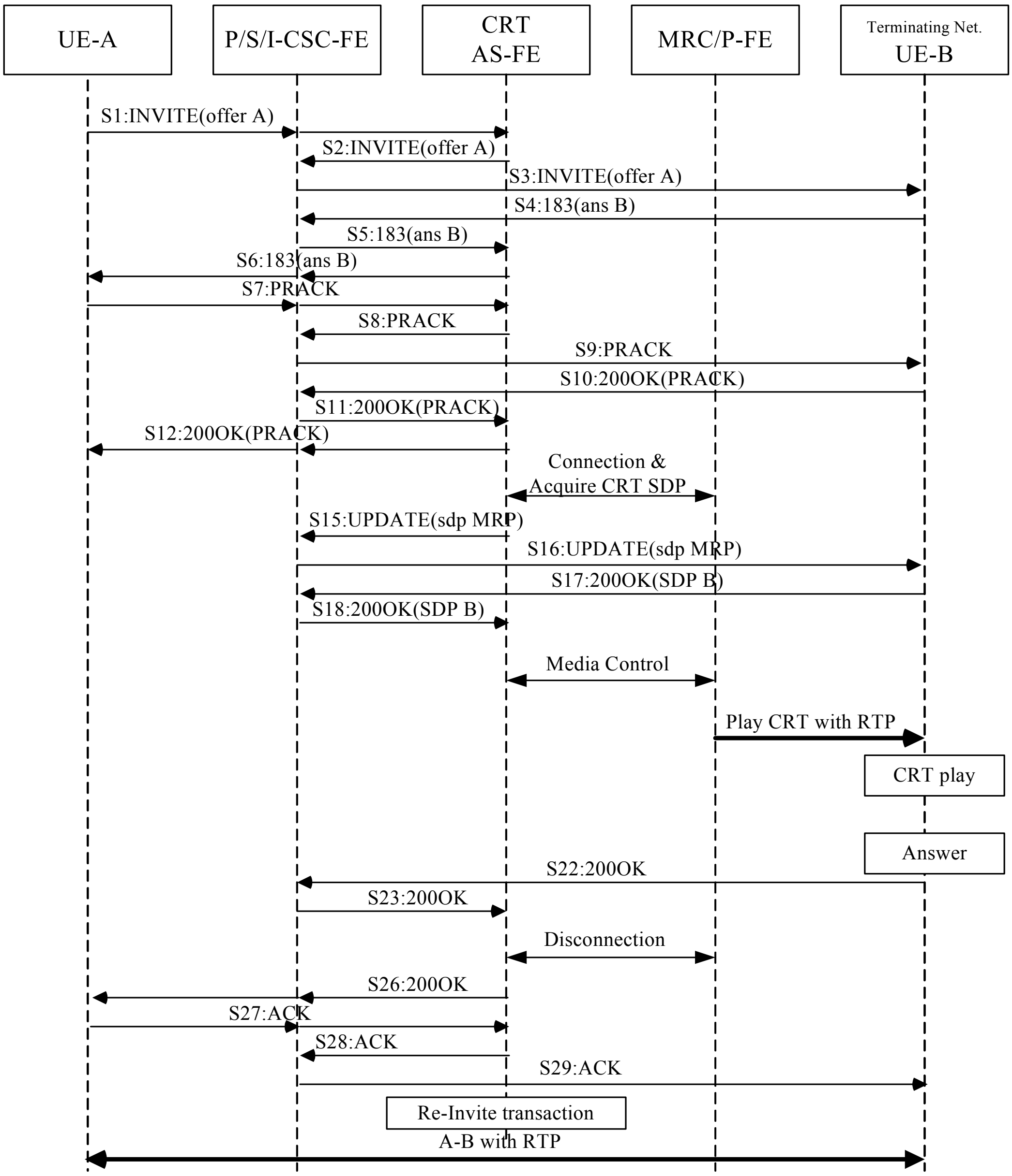

Figure I.1 – Signalling flows for CRT service using gateway model

The procedures of SIP signalling are as follows:

S1 INVITE (UE-A to AS-FE)

The INVITE request is sent from the UE-A to S-CSC-FE of the originating side. The S-CSC-FE evaluates the initial filter criteria. (The initial filter criteria identifies that the requested URI is subscribed to the CRT service.) If that matches, the S-CSC-FE will send the request toward the indicated CRT AS-FE. At this time, the contact value includes the UE-A's SIP URI that contains the IP address.

```
INVITE tel:+1-222-333-4444 SIP/2.0
Route: <sip:crt@as-fe.orig_ngn.net; lr>, <sip:1.23.233143.24@s-csc-fe.orig_ngn.net;lr>
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
P-Preferred-Identity: <sip:ue-a@orig_ngn.net>, <tel:+1-222-555-3333>
Privacy: none
Contact: <sip:192.100.200.51:5090>
Supported: timer, early-session
Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, MESSAGE, INFO, REFER, NOTIFY, SUBSCRIBE
Accept: application/sdp
Session-Expires: 3600;refresher=uac
Content-Type: application/sdp
…
```

S2-S3 INVITE (AS-FE to UE-B)

INVITE is sent toward the UE-B through network elements. The contact value includes the AS-FE's SIP URI.

```
INVITE sip:192.100.200.52:5080 SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=asfe171828
To: <tel:+1-222-333-4444>
Call-ID : asfed03a0sasfe0234123
CSeq: 101 INVITE
Supported : 100rel
Contact: <sip:192.100.100.100:5070>
Content-Type: application/sdp
…
```

S4-S6 183 (UE-B to AS-FE)

If 100rel option tag is presented in the 183 message with require/RSeq header SDP, it is sent toward the AS-FE through network elements. The Contact value includes the UE-B's SIP URI.

```
SIP/2.0 183 Session Progress
From: <sip:ue-a@orig_ngn.net>;tag=asfe171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : asfed03a0sasfe0234123
Require: 100rel
RSeq: 113
Cseq: 101 INVITE
Contact: <sip:192.100.200.52:5080>
Content-Type: application/sdp
…
```

S7 PRACK (UE-A to AS-FE)

UE-A is requested to send a PRACK request back, in order to acknowledge the provisional 183 (Session in Progress) response.

```
PRACK sip:192.100.100.100:5070 SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=asfe22222122
Call-ID : ueacb03a0suea00234123
RAck: 113 101 INVITE
Cseq: 102 PRACK
Content-Type: application/sdp
…
```

S8-S9 PRACK (AS-FE to UE-B)

PRACK is sent toward the UE-B through network elements.

```
PRACK sip:192.100.100.100:5070 SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=asfe171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : asfed03a0sasfe0234123
RAck: 113 101 INVITE
Cseq: 102 PRACK
Content-Type: application/sdp
…
```

S10-S12 200 OK (UE-B to UE-A)

The UE-B responds to the PRACK (S11) with a 200 OK. Then 200 OK is sent toward the UE-A through network elements.

```
SIP/2.0 200 OK
From: <sip:ue-a@orig_ngn.net>;tag=asfe171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : asfed03a0sasfe0234123
Cseq: 102 PRACK
…
```

S15-S16 UPDATE (AS-FE to UE-B)

AS-FE generates an UPDATE message with the SDP of the MRC-FE. UPDATE is sent toward the UE-B through network elements. The contact value includes the AS-FE's SIP URI.

```
UPDATE sip:192.100.200.51:5080 SIP/2.0
From: <sip:ue-a@ngn.net>;tag= asfe171828
To: <tel:+1-222-333-4444>;tag= 22222122
Call-ID : asfed03a0sasfe0234123
CSeq: 222 UPDATE
Contact: <sip:192.100.100.100:5070>
Content-Type: application/SDP
…
```

S17-S18 200 OK (UE-B to S-CSC-FE)

The UE-B responds to the UPDATE (S15) with a 200 OK including the SDP of the UE-A. Then 200 OK is sent toward the AS-FE through network elements. The contact value includes the UE-A's SIP URI.

```
SIP/2.0 200 OK
From: <sip:ue-a@ngn.net>;tag= asfe171828
To: <tel:+1-222-333-4444>;tag= 22222122
Call-ID : asfed03a0sasfe0234123
CSeq: 222 UPDATE
Contact: <sip:192.100.200.51:5090>
Content-Type: application/SDP
```

S22-S23 200 OK (UE-B to AS-FE)

When the called party answers, the UE-B sends a 200 OK final response to the INVITE request (S3). The S-CSC-FE forwards the 200 OK response to the AS-FE. The contact value includes the UE-B's SIP URI.

```
SIP/2.0 200 OK
From: <sip:ue-a@orig_ngn.net>;tag=asfe171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : asfed03a0sasfe0234123
CSeq: 101 INVITE
Contact: <sip:192.100.200.52:5080>
Content-Type: application/sdp
```

S25-26 200 OK (AS-FE to UE-A)

The S-CSC-FE forwards the 200 OK response to the UE-A. The contact value includes the AS-FE's SIP URI.

```
SIP/2.0 200 OK
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag= asfe22222122
Call-ID : ueacb03a0suea00234123
Contact: <sip:192.100.100.100:5070>
CSeq: 101 INVITE
…
```

S27-29 ACK (UE-A to UE-B)

The UE-A responds to the 200 OK with an ACK request to the UE-B. The UE-A and the UE-B should change the RTP session after re-INVITE transaction.

```
ACK sip:192.100.100.100:5070 SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag= asfe22222122
Call-ID : ueacb03a0suea00234123
CSeq: 101 INVITE
```

I.2 CRT service using application server model

Figure I.2 signalling flow shows a CRT service provided by the application server model scenario:

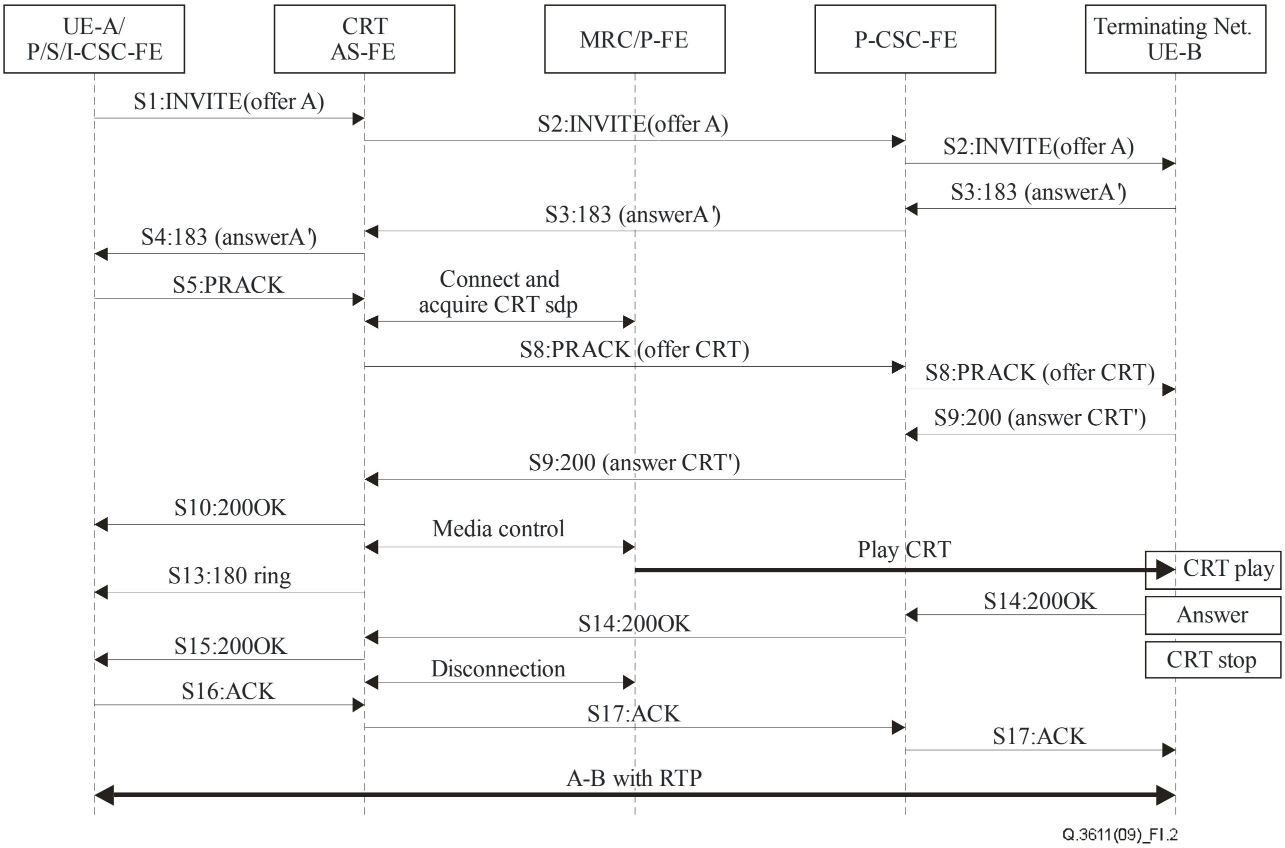

Figure I.2 – Signalling flows for CRT service using application server model

The procedures of SIP signalling are as follows:

S1 INVITE (originating side to S-CSC-FE)

The INVITE request is sent from the UE-A to the S-CSC-FE of the terminating side. The S-CSC-FE evaluates the initial filter criteria. (The initial filter criteria identifies that the requested URI is subscribed to the CRT service.) Therefore, the S-CSC-FE forwards the INVITE to the CRT AS-FE. The contact value includes the UE-A's SIP URI that contains the IP address.

```
INVITE sip:ue-b@term_ngn.net SIP/2.0
Route: <sip:crt@as-fe.term_ngn.net; lr>, <sip:1.23.233143.23@s-csc-fe.term_ngn.net;lr>
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
P-Preferred-Identity: <sip:ue-a@orig_ngn.net>, <tel:+1-222-555-3333>
Privacy: none
Contact: <sip:192.100.200.51:5090>
Supported: timer
Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, MESSAGE, INFO, REFER, NOTIFY, SUBSCRIBE
Accept: application/sdp
Session-Expires: 3600;refresher=uac
Content-Type: application/sdp
…
```

S2 INVITE (AS-FE to UE-B)

INVITE is sent toward the UE-B through network elements (AS-FE may be the Proxy server). The contact value includes the UE-A's SIP URI.

```
INVITE sip:192.100.200.52:5080 SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Supported: timer, early-session
Require:100rel, early-session
Contact: <sip:192.100.200.51:5090>
Content-Type: application/sdp
…
```

S3-S4 183 (UE-B to UE-A)

183 (Session Progress) is sent toward the AS-FE through network elements. The contact value includes the UE-B's SIP URI.

```
SIP/2.0 183 Session Progress
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Contact: <sip:192.100.200.52:5080>
Content-Type: application/sdp
…
```

S5 PRACK (UE-A to AS-FE)

PRACK is sent toward the AS-FE through network elements.

```
PRACK sip:ue-b@term_ngn.net SIP/2.0
From: <sip:crt@as-fe.term_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>; tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 PRACK
Contact: <sip: 192.100.200.51:5090>
Content-Length: 0
```

S8 PRACK (AS-FE to UE-B)

The AS-FE inserts an early-session SDP within PRACK request. PRACK is sent toward the UE-B through network elements.

```
PRACK sip:ue-b@term_ngn.net SIP/2.0
From: <sip:crt@as-fe.term_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>; tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 PRACK
Require: early-session
Contact: <sip: 192.100.200.51:5090>
Content-Type: application/sdp
Content-Disposition: early-session
…
```

S9 200 (UE-B to AS-FE)

200 (OK) is sent toward the AS-FE through network elements.

```
SIP/2.0 200 OK
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 PRACK
Contact: <sip:192.100.200.52:5080>
Content-Type: application/sdp
Content-Disposition: early-session
…
```

S10 200 OK (AS-FE to UE-A)

200 (OK) is sent toward the UE-A through network elements.

```
SIP/2.0 200 OK
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 PRACK
Contact: <sip:192.100.200.52:5080>
Content-Length: 0
```

S13 180 (AS-FE to UE-A)

180 (ringing) is sent toward the UE-A through network elements. The contact value includes the UE-B's SIP URI.

```
SIP/2.0 180 Ringing
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Contact: <sip:192.100.200.52:5080>
…
```

S14-S15 200 (UE-B to UE-A)

200 (OK) is sent toward the UE-A through network elements. The contact value includes the UE-B's SIP URI.

```
SIP/2.0 200 OK
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Contact: <sip:192.100.200.52:5080>
Content-Type: application/sdp
Content-Disposition: session
…
```

S16-S17 ACK (UE-A to UE-B)

The UE-A responds to the 200 OK with an ACK request to the UE-B.

```
ACK sip:192.100.200.52:5080 SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Content-Length: 0
```

I.3 CRT service using HTTP model

Figure I.3 signalling flow shows a CRT service provided by the HTTP model scenario:

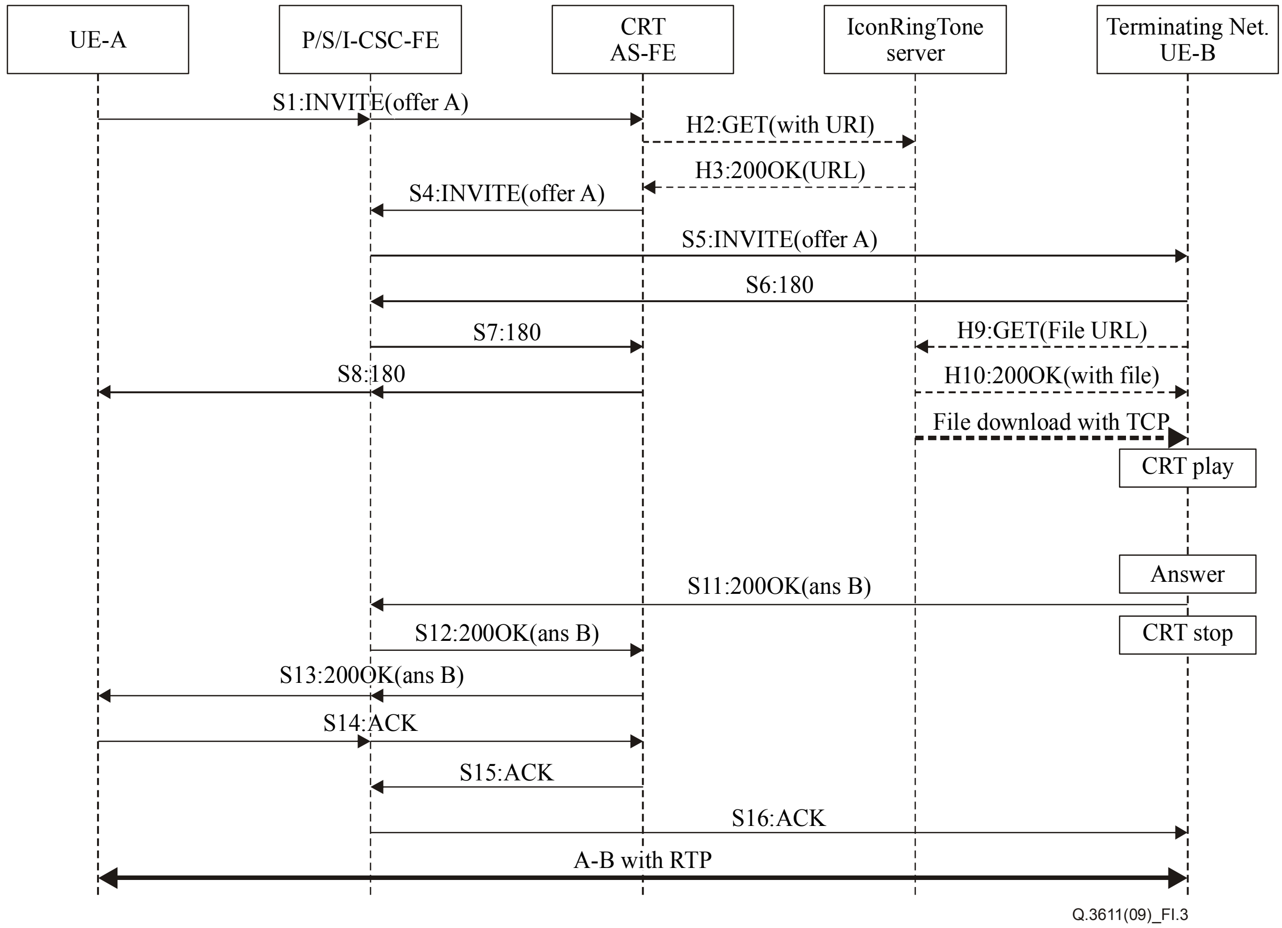

Figure I.3 – Signalling flows for CRT service using HTTP model

The procedures of SIP signalling are as follows:

S1 INVITE (UE-A to AS-FE)

The INVITE request is sent from the UE-A to the S-CSC-FE of the originating side. The S-CSC-FE evaluates the initial filter criteria (The initial filter criteria identifies that the requested URI is subscribed to the CRT service.). If that matches, the S-CSC-FE will send the request toward the indicated CRT AS-FE. At this time, the contact value includes the UE-A's SIP URI that contains the IP address.

```
INVITE tel:+1-222-333-4444 SIP/2.0
Route: <sip:crt@as-fe.orig_ngn.net; lr>, <sip:1.23.233143.24@s-csc-fe.orig_ngn.net;lr>
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
P-Preferred-Identity: <sip:ue-a@orig_ngn.net>, <tel:+1-222-555-3333>
Privacy: none
Contact: <sip:192.100.200.51:5090>
```

```
Supported: timer, early-session
Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, MESSAGE, INFO, REFER, NOTIFY, SUBSCRIBE
Accept: application/sdp
Session-Expires: 3600;refresher=uac
Content-Type: application/sdp
…
```

H2-H3 HTTP or SOAP (between AS-FE and CRT HTTP server)

The AS-FE gets CRT URL from the HTTP server using TCP-based protocol.

S4-S5 INVITE (AS-FE to UE-B)

The AS-FE inserts the Call-Info header or Alert-Info header within the INVITE request. The contact value includes the UE-A's SIP URI.

```
INVITE sip:192.100.200.52:5080 SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Contact: <sip:192.100.200.51:5090>
Content-Type: application/SDP
Call-Info : <http://crt_as.ngn.com/mmcid/bimg20001000000001731.gif>; purpose=icon,
            <http:// crt_as.ngn.com /mmcid/user_info_0251148012.xml>; purpose=info
Alert-Info : http://crt_as.ngn.com/mmcid/bimg20001000000001731.wav
…
```

S6-S7 180 (UE-B to AS-FE)

180 (ringing) is sent toward the UE-A through the network elements. The contact value includes the UE-B's SIP URI.

```
SIP/2.0 180 Ringing
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Contact: <sip:192.100.200.52:5080>
…
```

H9-H10 HTTP GET and File downloading (between AS-FE and CRT HTTP server)

The UE-B requests HTTP GET Message with URL. Then, the UE-B downloads and plays the CRT media file.

S11-S13 200 OK (UE-B to UE-A)

When the called party answers, the UE-B sends a 200 OK (with SDP B) response to the INVITE request. Then, the UE-B stops the CRT media play. The contact value includes the UE-B's SIP URI.

```
SIP/2.0 200 OK
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Contact: <sip:192.100.200.52:5080>
Content-Type: application/sdp
…
```

S14-S16 ACK (UE-A to UE-B)

The UE-A responds to the 200 OK with ACK request to the UE-B.

```
ACK sip:192.100.200.52:5080 SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
…
```

I.4 CRT service priority using application server model

Figure I.4 signalling flow shows an IMS-based CRT service priority provided by the application server model where the terminating side has higher priority.

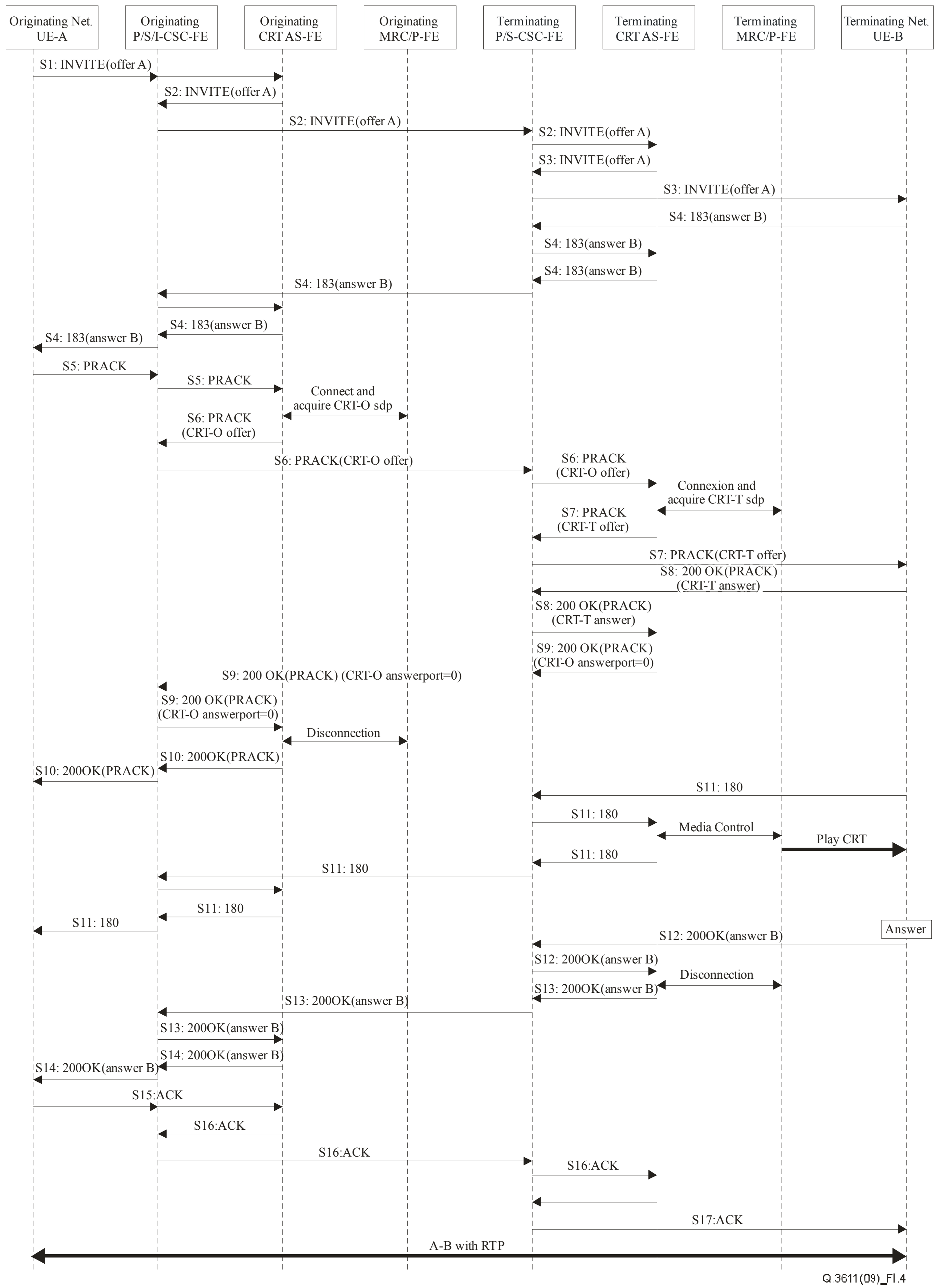

Figure I.4 – Signalling flows for CRT service with priority using application server model

The procedures of SIP signalling are as follows:

S1-S3 INVITE (UE-A toward UE B)

The INVITE request is sent from the UE-A to the S-CSC-FE of the originating side. The originating S-CSC-FE evaluates the initial filter criteria and forwards the INVITE to the originating CRT AS-FE. AS-FE forwards the INVITE to the terminating S-CSC-FE, and the terminating S-CSC-FE evaluates the initial filter criteria and forwards the INVITE to the terminating CRT AS-FE, the terminating CRT AS-FE checks the CRT priority setting, then the terminating CRT finally forwards the INVITE to the UE-B. The contact value includes the UE-A's SIP URI that contains the IP address. The "early-session" and "100rel" option-tag is added by the CRT AS-FE into the Require header of the INVITE request.

```
INVITE sip:ue-b@term_ngn.net SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
P-Preferred-Identity: <sip:ue-a@orig_ngn.net>, <tel:+1-222-555-3333>
Privacy: none
Contact: <sip:192.100.200.51:5090>
Supported: timer, early-session, 100rel
Require:100rel,early-session
Accept: application/sdp
Session-Expires: 3600;refresher=uac
Content-Type: application/sdp
…
```

S4 183 (UE-B toward UE-A)

The UE B responds with "183 Session Progress" towards the terminating CRT AS-FE which indicates the UE-B supporting the early session.

```
SIP/2.0 183 Session Progress
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Contact: <sip:192.100.200.52:5080>
Content-Type: application/sdp
…
```

S5 PRACK (UE-A toward originating CRT AS-FE)

The UE-A sends PRACK of the 183 response to the originating CRT AS-FE.

```
PRACK sip:ue-b@term_ngn.net SIP/2.0
From: <sip:crbt@as-fe.term_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>; tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 PRACK
Contact: <sip: 192.100.200.51:5090>
```

S6 PRACK (originating CRT AS-FE toward terminating CRT AS-FE)

The originating CRT AS-FE adds the CRT-O offer SDP in the PRACK message and forwards the PRACK message to the terminating CRT AS-FE.

```
PRACK sip:ue-b@term_ngn.net SIP/2.0
From: <sip:crbt@as-fe.term_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>; tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 PRACK
Contact: <sip: 192.100.200.51:5090>
Content-Type: application/sdp
Content-Disposition: early-session
Content-Length: (…)

v=0
o=- 2987933616 2987933616 IN IP4 aaa:bbb:eee:fff
s=-
c=IN IP4 ccc:aaa:bbb:acc
t=0 0
m=audio 3456 RTP/AVP 97
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes
m=video 3400 RTP/AVP 98
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
```

S7 PRACK (terminating CRT AS-FE toward UE-B)

Due to the fact that the terminating CRT has priority according to the priority setting, the terminating CRT AS-FE adds CRT-T offer SDP, instead of CRT-O offer, in a PRACK request and forwards the PRACK to the UE-B.

```
PRACK sip:ue-b@term_ngn.net SIP/2.0
From: <sip:crbt@as-fe.term_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>; tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 PRACK
Contact: <sip: 192.100.200.51:5090>
Content-Type: application/sdp
Content-Disposition: early-session
Content-Length: (…)

v=0
o=- 2987933616 2987933616 IN IP4 eee:fff:aaa:bbb
s=-
c=IN IP4 ccc:aaa:bbb:acc
t=0 0
m=audio 3456 RTP/AVP 97
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes
m=video 3400 RTP/AVP 98
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
```

S8 200 OK (UE-B toward terminating CRT AS-FE)

The UE-B completes the terminating CRT media negotiation and sends 200 OK of PRACK with a CRT-T answer SDP to the terminating CRT AS-FE. The contact value includes the UE-B's SIP URI.

```
SIP/2.0 200 OK
From: <sip:crbt@as-fe.term_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>; tag=22222122
Call-ID : ueacb03a0suea00234123
```

```
CSeq: 1 PRACK
Contact: <sip: 192.100.200.52:5080>
Content-Type: application/sdp
Content-Disposition: early-session
Content-Length: (…)

v=0
o=- 2987933616 2987933616 IN IP4 aaa:bbb:ccc:ddd
s=-
c=IN IP4 aaa:bbb:ccc:ddd
t=0 0
m=audio 3466 RTP/AVP 97
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes…
m=video 3400 RTP/AVP 98
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
```

S9 200 OK (terminating CRT AS-FE toward originating CRT AS-FE)

The terminating CRT AS-FE adds the CRT-O answer SDP setting port to refuse the originating CRT media negotiation in the 200 OK response, instead of the previous CRT-T answer SDP, and forwards the 200 OK to the originating CRT AS-FE.

```
SIP/2.0 200 OK
From: <sip:crbt@as-fe.term_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>; tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 PRACK
Contact: <sip: 192.100.200.52:5080>
Content-Type: application/sdp
Content-Disposition: early-session
Content-Length: (…)

v=0
o=- 2987933616 2987933616 IN IP4 aaa:bbb:ccc:ddd
```

```
s=-
c=IN IP4 aaa:bbb:ccc:ddd
t=0 0
m=audio 0 RTP/AVP 97
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes…
m=video 0 RTP/AVP 98
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
```

S10 200 OK (originating CRT AS-FE toward UE-A)

The originating CRT AS-FE removes the CRT-O answer SDP and forwards the 200 OK to the UE-A.

```
SIP/2.0 200 OK
From: <sip:crbt@as-fe.term_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>; tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 PRACK
Contact: <sip: 192.100.200.52:5080>
```

S11 180 (UE-B toward UE-A)

The UE-B sends a 180 Ring response to the UE-A through network elements. When the terminating CRT AS-FE receives the response, the terminating CRT AS-FE plays the terminating CRT to the UE-B.

```
SIP/2.0 180 Ringing
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Require :100rel
Contact: <sip:192.100.200.52:5080>
```

S12-14 200 (UE-B toward UE-A)

200 (OK) is sent toward the UE-A through network elements. The contact value includes the UE-B's SIP URI.

```
SIP/2.0 200 OK
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Contact: <sip:192.100.200.52:5080>
Content-Type: application/sdp
Content-Disposition: session
…
```

S15-17 ACK (UE-A toward UE-B)

The UE-A responds to the 200 OK with an ACK request to the UE-B.

```
ACK sip:192.100.200.52:5080 SIP/2.0
From: <sip:ue-a@orig_ngn.net>;tag=171828
To: <tel:+1-222-333-4444>;tag=22222122
Call-ID : ueacb03a0suea00234123
CSeq: 1 INVITE
Content-Length: 0
```

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