ATIS STANDARD

ATIS-1000068

Support of TTY Service Over IP Using Global Text Telephony

TECHNICAL REPORT

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ATIS-1000068

Technical Report on

Support of TTY Service Over IP Using Global Text Telephony

Alliance for Telecommunications Industry Solutions

Approved August 3, 2017

Abstract

This Technical Report (TR) describes the means by which the Teletypewriter (TTY) service can be provided over Internet Protocol (IP) between operators' networks through the use of the Global Text Telephony (GTT) capability which enables simultaneous audio and/or video with text media stream.

Foreword

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between providers, customers, and manufacturers. The Packet Technologies and Systems Committee (PTSC) develops and recommends standards and technical reports related to services, architectures, and signaling, in addition to related subjects under consideration in other North American and international standards bodies. PTSC coordinates and develops standards and technical reports relevant to telecommunications networks in the U.S., reviews and prepares contributions on such matters for submission to U.S. ITU-T and U.S. ITU-R Study Groups or other standards organizations, and reviews for acceptability or per contra the positions of other countries in related standards development and takes or recommends appropriate actions.

The mandatory requirements are designated by the word *shall* and recommendations by the word *should*. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages. The word *may* denotes an optional capability that could augment the standard. The standard is fully functional without the incorporation of this optional capability.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, PTSC, 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, PTSC, which was responsible for its development, had the following leadership:

M. Dolly, PTSC Chair (AT&T)

V. Shaikh, PTSC Vice Chair (Vencore Labs)

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1 Introduction

This Technical Report (TR) describes the means by which the Teletypewriter (TTY) service can be provided over IP between operators' networks through the use of the Global Text Telephony (GTT) capability which enables simultaneous audio and/or video with text media stream.

TTY service allows real time conversation in text between two persons having a Baudot-capable device. This service is supported through the Circuit Switched (CS) public network. Although new Internet technologies have reduced the need for this service, it still plays an important role, especially for emergency 9-1-1 calls.

Real-Time Text (RTT) is a term used to define the ability to instantly communicate text as it is typed, as opposed to after a sentence or thought is completed, in the manner of instant messaging. This term has now been replaced with Fast Text. RTT can now be signalled over IP networks. RTT is combined with voice and can be optionally combined with video. When this combined service is provided by an IP Multimedia Subsystem (IMS) network, it is referred to as GTT. GTT is supported in IMS networks via RTT capability using IETF SIP/SDP for the negotiation of the text media stream and IETF RFC 4103 [RFC 4103] RTP-text for transport, with text coded according to ITU-T Recommendation T.140 [T.140].

Figure 1.1 shows a generalized view of the GTT feature architecture. It combines different networks and network types and integrates text conversation systems already existing within these networks.

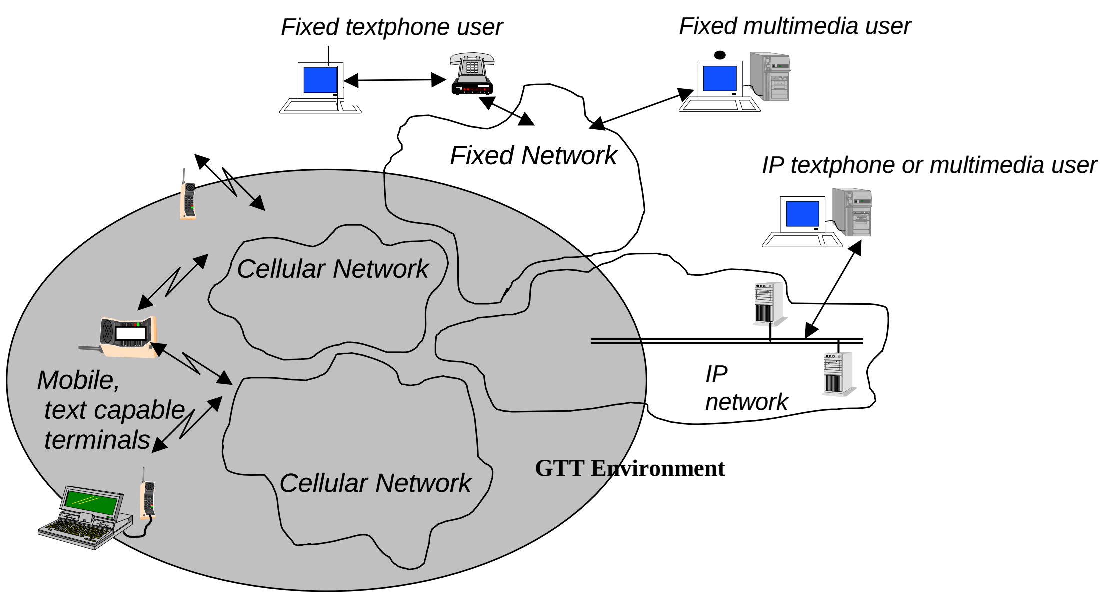

Figure 1.1 – Generalized GTT Architecture

Throughout the remainder of this document the following terms are used as described below:

- TTY – this term is used in reference to a real-time text service where the transport of text for this service is technology dependent: i.e., Baudot for circuit and RFC 4103 [RFC 4103] text for packet.
- GTT – this term is used to refer to the ability to support audio and/or video streams simultaneously with text (RTT) in an IP network.

- RTT – this term is used to refer to the text stream in an IP network as encoded per RFC 4103 [RFC 4103].

2 Abbreviations

ADA	Americans with Disabilities Act
ALG	Application Level Gateway
AS	Application Server
ATIS	Alliance for Telecommunications Industry Solutions
AVP	Audio Visual Profile
AVPF	Audio Visual Profile with Feedback
BGCF	Border Gateway Control Function
CS	Circuit Switched
CTM	Cellular Text Telephone Modem
E-CSCF	Emergency Call Session Control Function
EDT	European Deaf Telephone
IBCF	Interconnection Border Control Function
IETF	Internet Engineering Task Force
ITU-T	International Telecommunication Union - Telephony
GTT	Global Text Telephony
IM-MGW	Internet Protocol-Multimedia Media Gateway
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IWF	Interworking Function
LNG	Legacy Network Gateway
LPG	Legacy PSAP Gateway
LRF	Location Retrieval Function
LS	Location Server
MGCF	Media Gateway Control Function
MGW	Media Gateway
MRFC	Media Resource Function Controller
MRFP	Media Resource Function Processor
P-CSCF	Proxy Call Session Control Function
PCM	Pulse Code Modulation
PCMU	Pulse Code Modulation mu-law
POTS	Plain Old Telephone Service
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
RDF	Routing Determination Function
RTT	Real-Time Text

S-CSCF	Serving Call Session Control Function
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TDM	Time Division Multiplexed
TN	Telephone Number
TTY	Teletypewriter
UDP	User Datagram Protocol
VoIP	Voice over Internet Protocol
VoLTE	Voice over Long Term Evolution

3 Support of TTY Using Global Text Telephony (GTT)

3.1 Overview

The support of the TTY service is a regulatory requirement for North American public networks. As these networks migrate to packet technologies, the TTY service must also be migrated to IP. GTT offers real time conversation in text, optionally combined with voice and/or video. GTT is mainly used for distant conversation with hearing or speech impaired users.

On the PSTN, different systems for text telephony exist and are used in different regions, e.g., Baudot (in US), or EDT, V.21, Bell103, Minitel, and V.18 in other countries. They all use different modem technologies within PCM and different character coding for the transmission of text. They are described in the annexes of ITU-T Recommendation V.18 [V.18]. Baudot is a protocol used to signal a limited set of uni-case letters and figures represented by five bit codes at a rate of 45 baud in the U.S. No error correction is provided. Any party of a GTT call may at any time initiate text or send voice. Speech and text may be used in an alternating manner during a conversation on the PSTN. It is also possible that speech is transferred in one direction and text in the opposite direction. However, speech and text cannot be used in the same direction at the same time.

In the 3G radio interface, a dedicated CTM modem is used (see 3GPP TS 26.226 [TS 26.226]), which is terminated within the CS domain and interworked to PSTN using an in-band text telephony format.

A generic Interworking Function (IWF) can be introduced into the call whenever a conversion is needed between different text encoding, e.g., between RTT and Baudot. Interworking between RTT over RTP from IP networks with CS network text telephony is provided by the MGCF and IM-MGW or IMS triggering the insertion of an interworking (conversion) function. The interworking capabilities in the MGW support the detection of Baudot on the CS side and the conversion between Baudot and RTT over RTP.

Interworking between RTT over RTP and Baudot may also be necessary with other packet networks or other components within the same IP network not supporting RTT. In this case, the IWF may be placed closer to the access interface point. This same approach may be used as an alternate solution for interworking with CS networks thus avoiding the introduction of RTT interworking.

The procedures to detect and convert text/modem involve valuable MGW resources. The GTT interworking procedures specified in this document allow more efficient use of MGW resources by focusing the application of text/modem conversion on the minority of calls that use text telephony.

It is assumed that SIP terminals supporting text media will not automatically offer text media, but that this will be instead governed by SIP terminal configuration options and user interactions to suit the communication preferences and abilities of the user. However, a SIP terminal desiring to set up a GTT call will offer RTT media ***(along with voice media),*** possibly with video media. The interworking function then provides the conversion between RTT over RTP and text/modem signals. Conversely, if the SIP terminal does not request RTT support, no Interworking function is necessary. An IMS Multimedia SIP terminal configured to use RTT Telephony but receiving an SDP offer for voice-only media will accept this offer and then send its own subsequent SDP offer adding text media. When receiving such a subsequent offer for text media, the MGW will provide the conversion between RTT over RTP and text/modem signals at the CS interface. If the terminating mobile device does not offer RTT, no interworking function is necessary.

The text in RTT is coded in a common presentation protocol, ITU-T Recommendation T.140 [T.140]. If necessary this presentation protocol will be converted to or from any legacy mode character code used in other networks.

By using the described GTT functions, a real-time text conversation session can be conducted between GTT- supported mobile text capable terminals. Different terminal function combinations and GTT host environments give different opportunities regarding combinations of text with voice and video. Valid combinations are:

- Alternating text and voice,
- Simultaneous text and voice,
- Simultaneous text, video, and voice.

3.2 Control Plane

A SIP terminal should offer Audio Visual Profile (AVP) for all media streams containing text. Audio Visual Profile with Feedback (AVPF) shall only be used in cases where there is an explicit demand for the AVPF RTCP reporting timing or feedback messages.

A SIP terminal configured to automatically enable RTT (e.g., because the SIP terminal is used by a deaf or hearing-impaired person or a person wanting to communicate with such an impaired person) shall accept an initial INVITE request for a SIP dialogue if the SDP offer does not include RTT media. It shall then send a new SDP offer (e.g., in a SIP UPDATE request during call establishment) adding text media for RTT conversation.

NOTE: As one example, incoming calls from a PSTN interworked by an IWF may not contain media for real time text conversation in the initial SDP offer. The new offer adding media for real time text conversation enables the transport of RTT towards the SIP terminal.

Below is an example of SDP that shows the use of RTT with T.140 encoding:

```
m=text 11000 RTP/AVP 98
a=rtpmap:98 t140/1000
```

In order to avoid lost data, it is recommended that some form of redundancy be provided. The default method for the support of redundancy is through the use of IETF RFC 2198 [RFC 2198] procedures that allow for the transmission of the original text plus redundant data.

The following is an example showing the optional use of redundancy per IETF RFC 2198 [RFC 2198] showing the use of two levels of redundancy.

```
m=text 11000 RTP/AVP 100 98
a=rtpmap:98 t140/1000
a=rtpmap:100 red/1000
a=fmtp:100 98/98/98
```

Optionally, when text media is supported by a device, a “text” media feature tag may be included in the Contact header field during registration and call establishment by devices and components capable of supporting RTT. The presence of this tag allows the inclusion of an IWF. The “sip.text” media feature tag is defined by IETF RFC 3840 [RFC 3840] as a “base” tag. It is added to the Contact header field with neither the “+” nor the “sip.” prefix (simply “text”).

3.3 User Plane

Use of the redundancy coding variant specified in IETF RFC 4103 [RFC 4103] is recommended for error resilience.

RTT media type RTP payload format for ITU-T Recommendation T.140 is specified in [T.140]. Redundant transmission provided by the RTP payload format is recommended in error prone channels.

The RTT stream is defined as T.140/RTP/UDP/IP:

T.140
RTP
UDP
IP

3.4 GTT Procedures

3.4.1 General

Two GTT-capable devices will be able to interconnect using standard SIP signalling and SDP offer/answer procedures. The resulting established session will consist of one audio and one RTT stream and optionally separate video stream. Since the RTT stream is separate from the audio and video streams, it is possible that the users are able to communicate by switching between text and audio/video or simultaneously using text and/or audio/video.

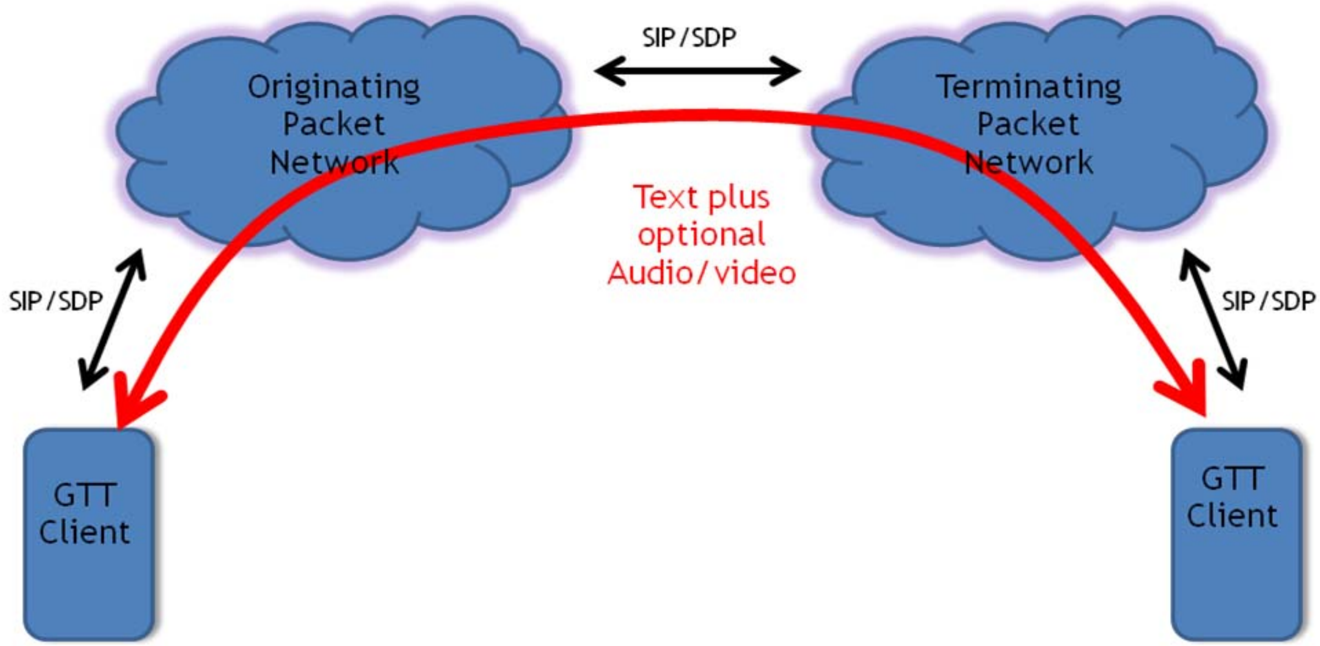

Figure 3.1 – Interconnecting GTT Clients

When communicating between a GTT-capable device and the Baudot device in the PSTN, it is necessary to convert the text and audio stream into PCM Baudot and audio. The packet RTT will be converted to Baudot tones. This interworking functionality is provided by the MGW component at the border between the IP and circuit networks.

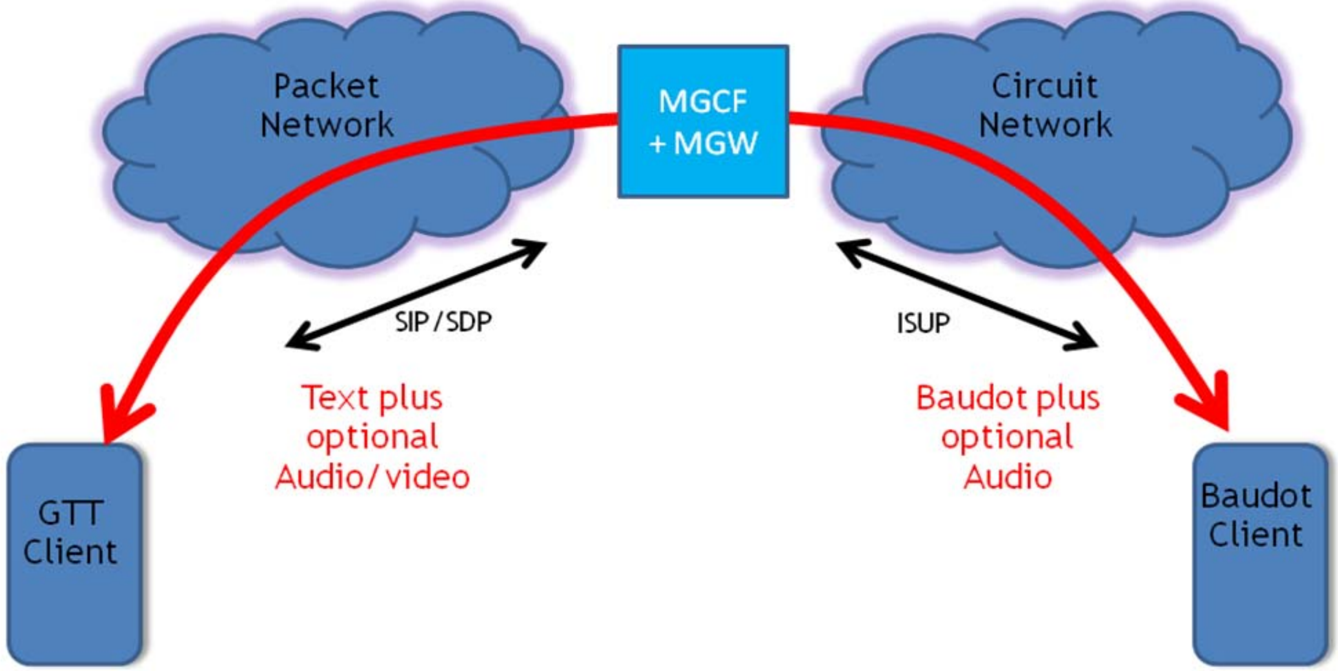

Figure 3.2 – Interworking between GTT Client and Circuit Based Client

In the event that a GTT-capable device needs to establish a call with the PSTN, but the MGCF does not support the conversion to GTT, or the GTT-capable device is calling a device in another IP network that does not support GTT, then the VoIP network needs to provide an alternative means to convert between RTT and the legacy Baudot tone. This conversion can be provided by an IWF within the IP network hosting the GTT-capable device.

The following two figures show two possible network architectures that may be used to provide the IWF functionality within the IP network. Other architectures may be possible. These figures illustrate a connection toward the PSTN. However, similar connections to other IP networks through an IBCF are possible, but not shown.

The first example architecture follows the standard IMS based Application Server (AS) model. In this case, an AS is looped into the call path by the S-CSCF. The AS may then communicate with an MRFC/MRFP that provides media termination points between the two incompatible end points. This allows for the conversion between RTT and Baudot tones.

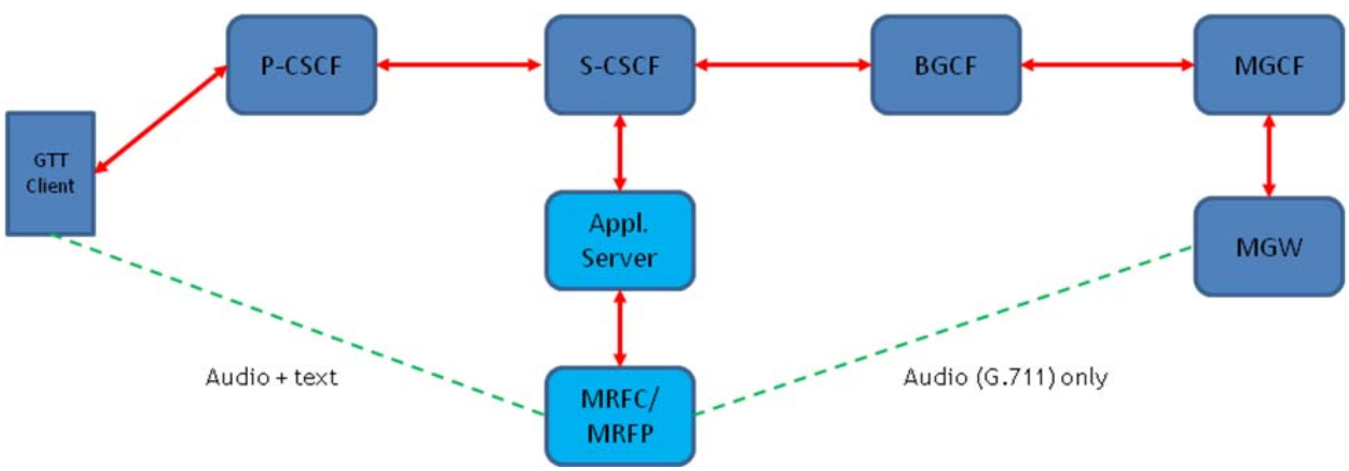

Figure 3.3 – Interworking Based on IMS Application Server Model

In the second example architecture, an IMS Application Level Gateway (ALG) can provide the same conversion functionality. The IMS ALG is typically hosted by P-CSCF.

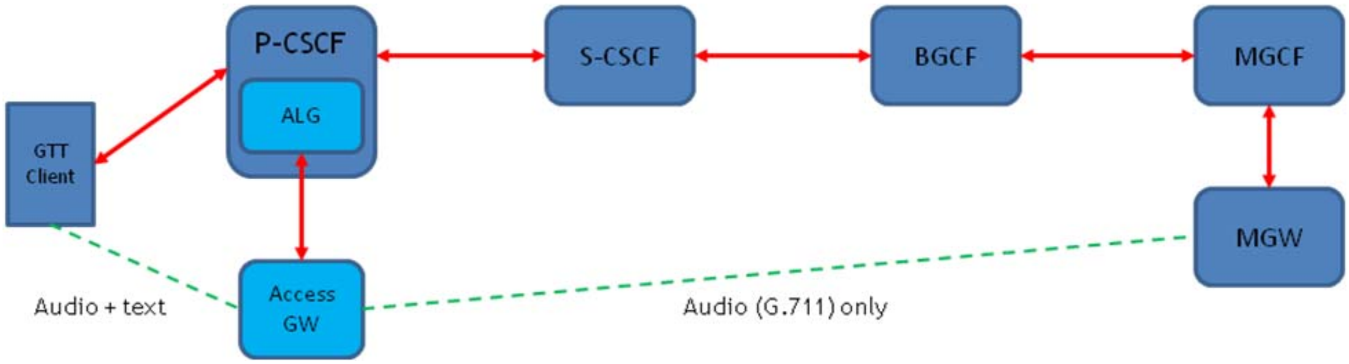

Figure 3.4 – Interworking Based on IMS ALG

The call flows provided later within this document use the generic label of “IWF” to represent any implementation supporting this conversion functionality.

3.4.2 Originating GTT Calls

When originating calls, if the user wishes to immediately offer RTT, the initial INVITE can be sent including both audio and text media types; video may optionally also be included. Assuming the terminating party also supports and accepts the offer with RTT, it will return the SDP answer that includes the text media type and other accepted media types.

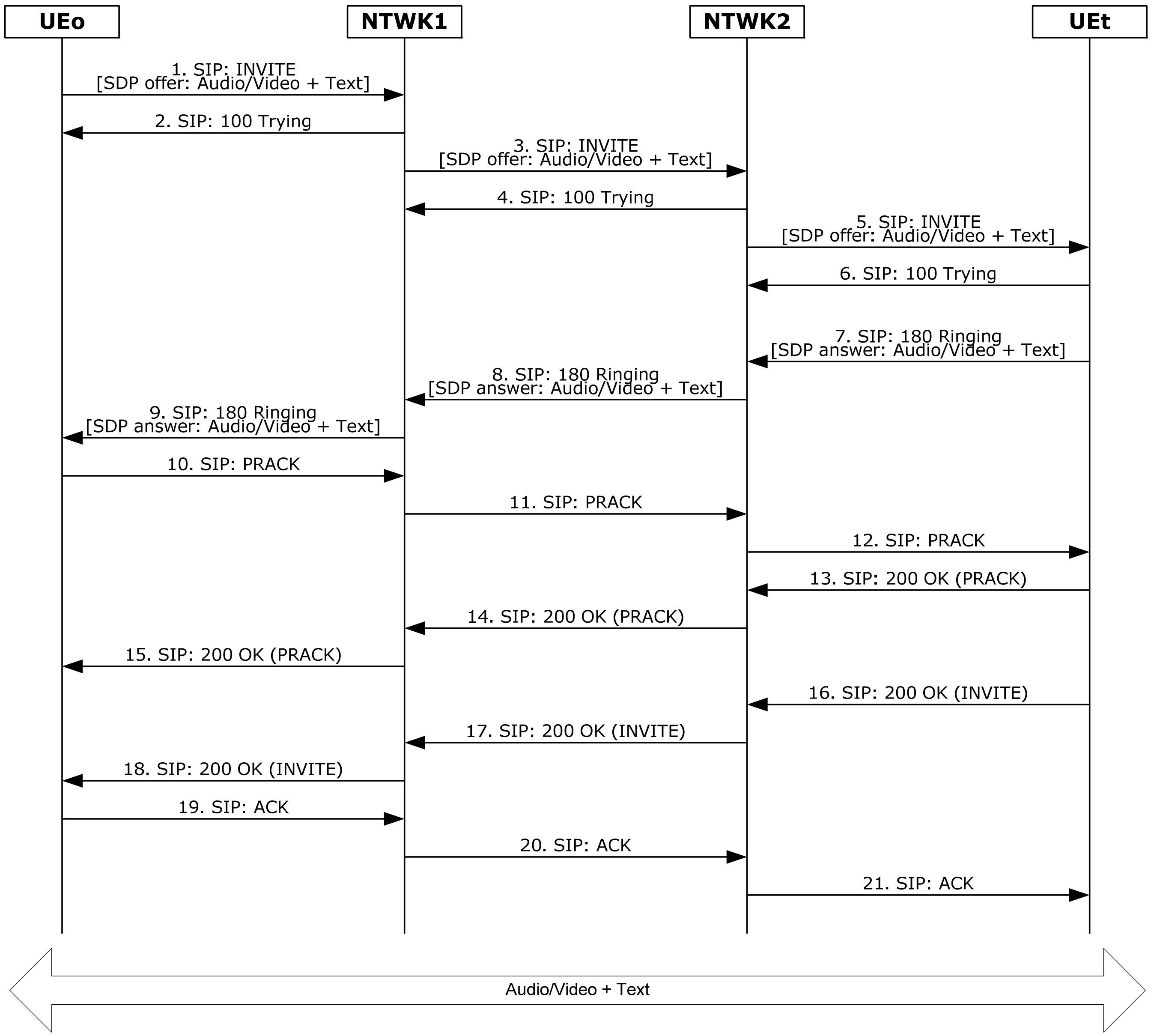

Figure 3.5 – GTT Call Flow with Text in Initial Offer

If the terminating party does not support RTT or wishes not to use it, it can accept the audio/video stream of the offer and reject just the text media stream by setting the port of the Text stream to zero. The call can continue successfully using only audio/video.

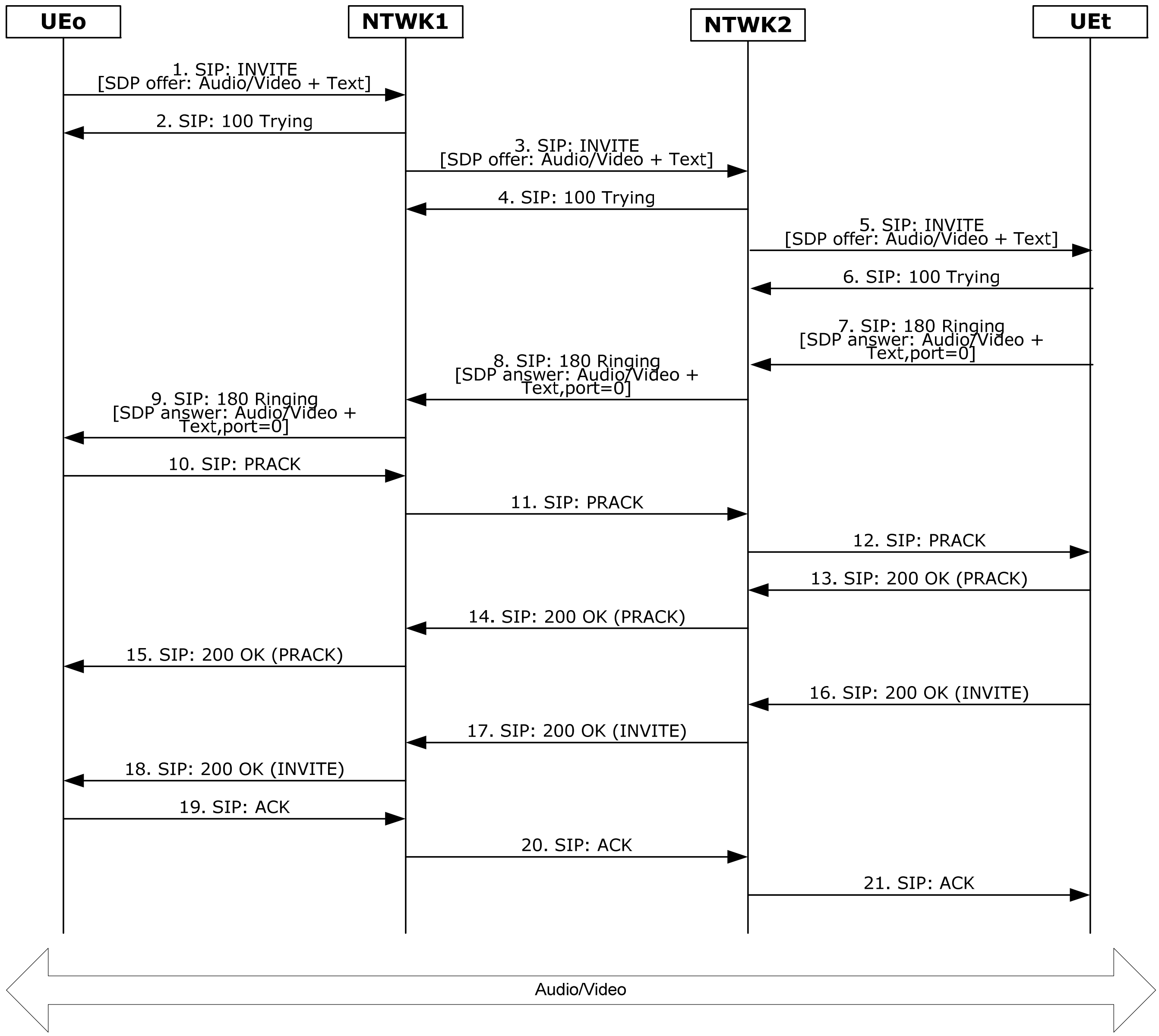

Figure 3.6 – GTT Call Flow with Text Initial Offer Rejected

If the originating party does not know if the terminating party supports RTT or if they do not want to initially offer RTT, the originating party can send the initial INVITE request with an offer containing only audio/video media streams. If upon receiving the incoming call, the terminating party wishes to add text media, it must first accept the initial offer and then send a subsequent offer adding the new text media stream.

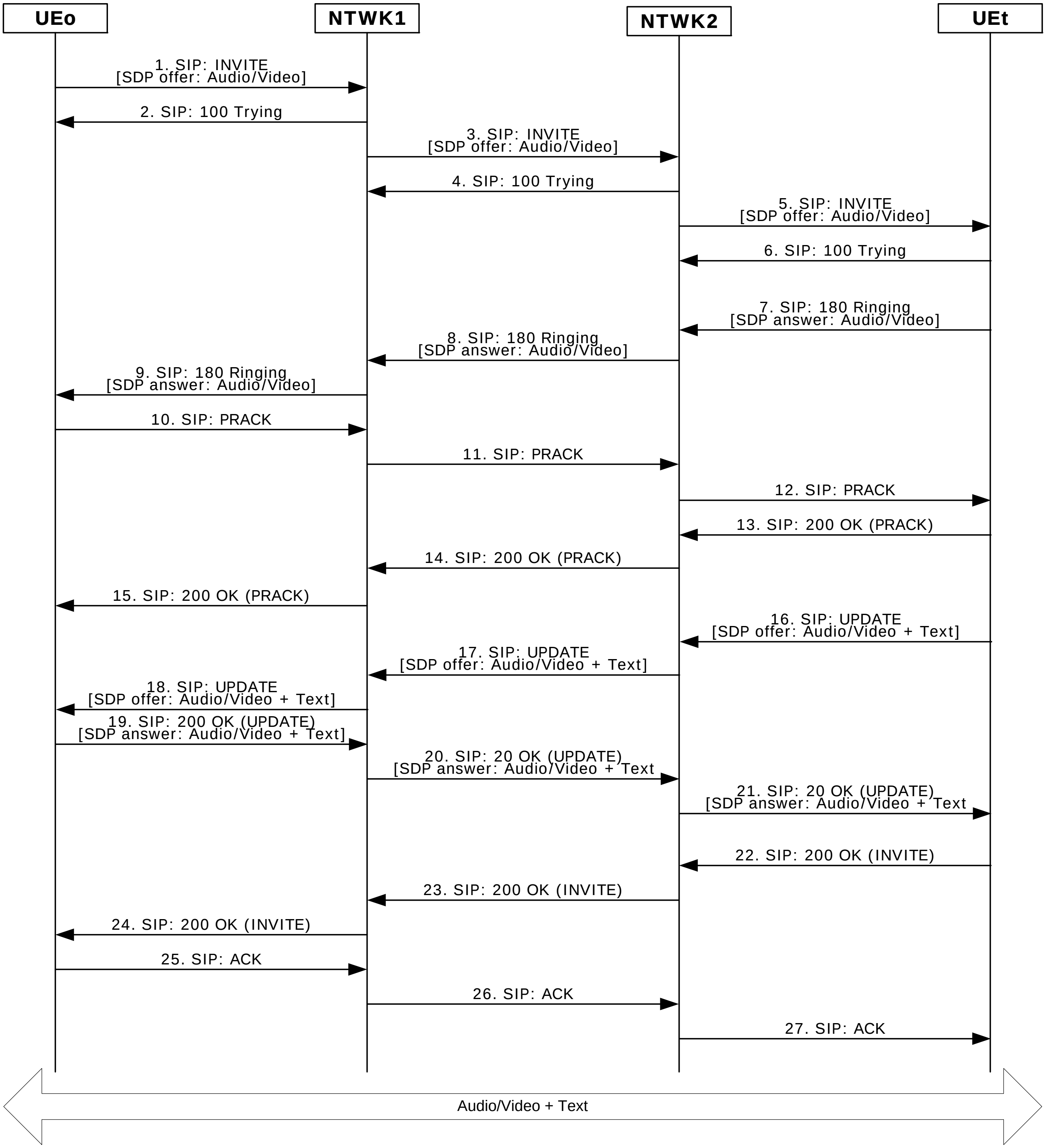

Figure 3.7 – GTT Call Flow with Text in Subsequent Offer

3.4.3 Incoming Call Interworking

When starting the (non-emergency) session setup signaling from a CS based network towards an IP network, the IWF in NTWK2 has no knowledge whether the call will attempt to use text telephony.

The IWF offers only audio media when setting up a call towards the SIP terminal and waits for the SIP terminal desiring RTT media to send a new offer adding RTT media attribute prior to inserting an Interworking function in the MGW.

The following shows an example call flow.

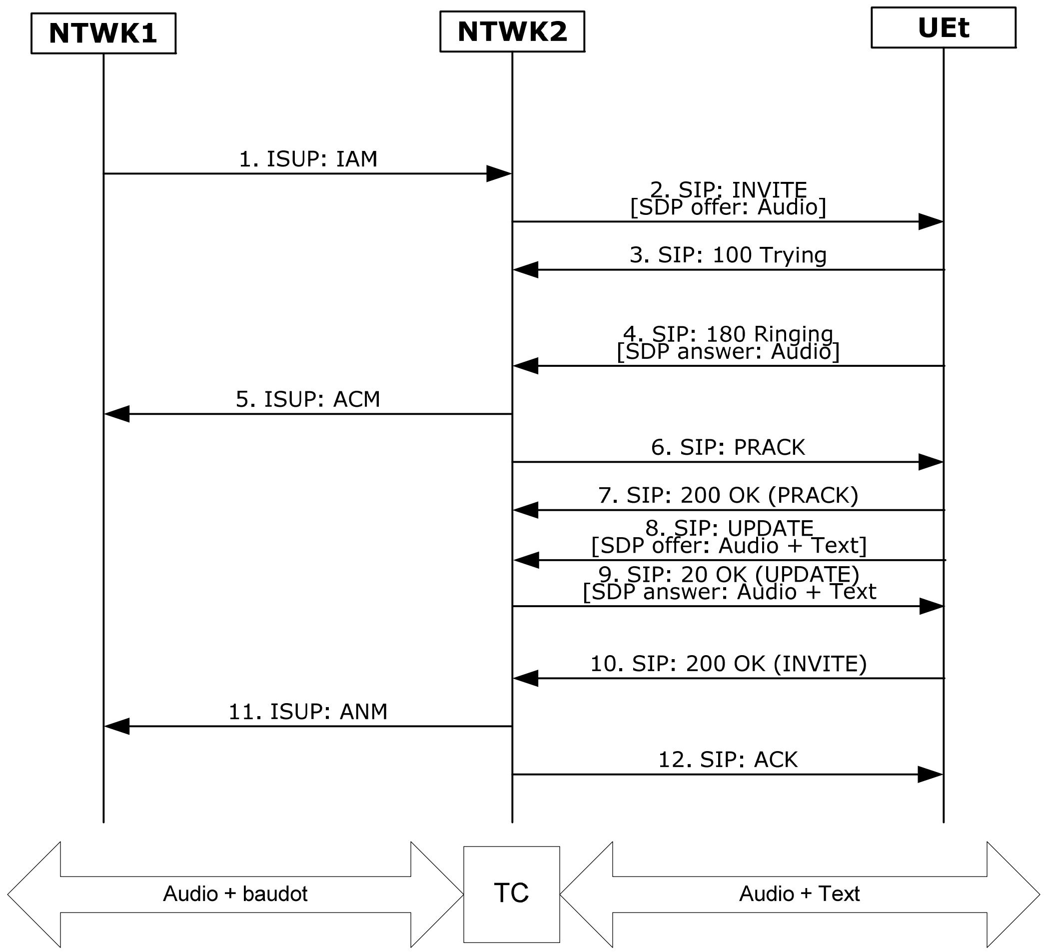

Figure 3.8 – CS Originated Session - Initial INVITE Offering Audio Only

Upon receipt of an IAM request for a speech or 3.1 kHz audio call, the IWF (e.g., MGCF and IM-MGW) starts the call setup by sending an INVITE request offering audio media applying the interworking procedures.

SIP terminals supporting RTT and configured to use it will send a new SDP offer including an audio and a RTT media line within a subsequent UPDATE prior to answer or re-INVITE request after answer.

When RTT interworking between IP and CS networks is required, the IWF shall reserve corresponding RTT media resources in the MGW and thereby request the insertion of the Interworking function, and if resources are available, return an SDP answer with audio and RTT media attributes.

3.4.4 Outgoing Call Interworking

Figure 3.9 shows an example call flow where the SIP terminal requests RTT by sending an SDP offer including one audio line and one text media line within an initial INVITE message.

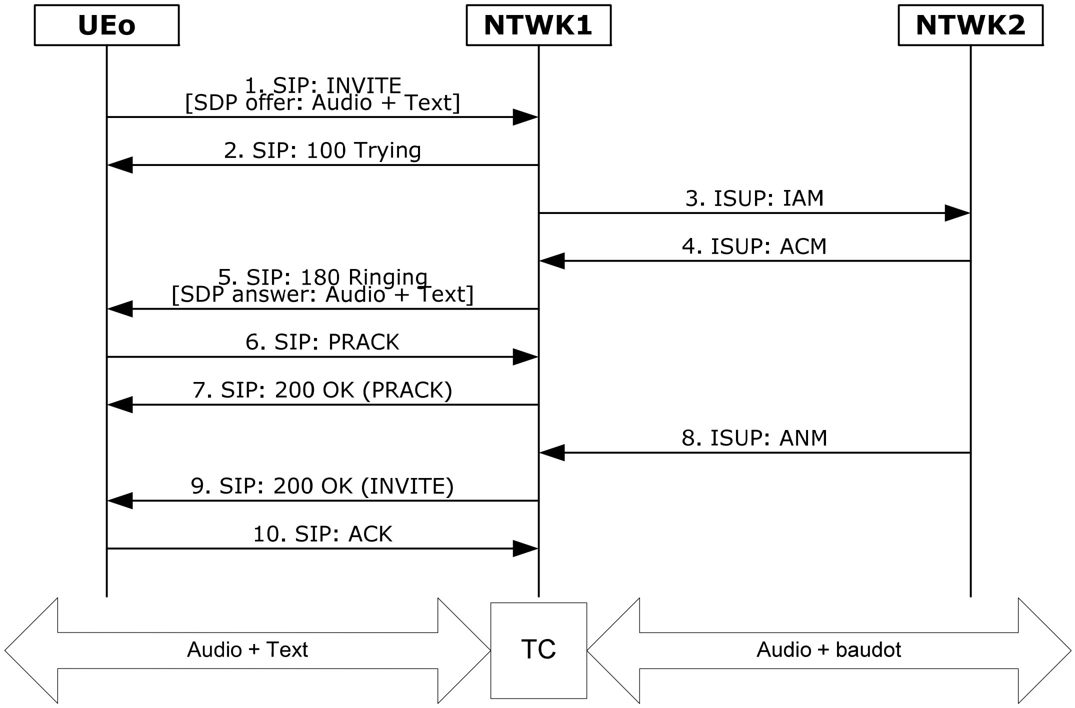

Figure 3.9 – SIP Terminal Originated Session - Initial INVITE Offering Audio and Text

Upon receipt of a SIP INVITE request offering audio and text media, the IWF starts the call setup at the CS side by sending an IAM requesting a speech or 3.1 kHz bearer, and completes the call setup on the IP and CS sides, returning an SDP answer that includes audio and RTT media.

The IWF triggers the insertion of an Interworking function in the MGW for the duration of the call if a RTT media stream is established.

The IWF reserves corresponding RTT media resources in the MGW and activates the Interworking function, and if resources are available, returns an SDP answer with audio and RTT.

3.4.5 Subsequent SDP Offer/Answer Exchange Adding Text to an Existing Session

If only audio and/or video media has been offered in the initial SDP offer, the SIP terminal can also request GTT support by sending a new SDP offer including audio/video and RTT when a SIP dialogue (early or confirmed) has already been established.

The IWF will then be triggered to provide the conversion.

3.5 GTT Procedures Utilizing Text Media Feature Tag

When a party has multiple devices registered under the same number, e.g., a residential gateway supporting an RJ11-attached POTS phone (no RTT support) and a Wi-Fi-attached tablet with an RTT-capable softphone, the presence of the RTT tag ("text" as defined by IETF RFC 3840 [RFC 3840]) during the SDP offer/answer exchange will permit the network to selectively direct session offers featuring RTT to the RTT-capable endpoint (either exclusively or in conjunction with the POTS phone). If a tag was not provided during the SDP offer/answer exchange, this would not be possible.

- For example: Bob has VoIP service with both a POTS phone gateway (RJ11 jack) and a Wi-Fi tablet connected and registered. The tablet is running an RTT client. Voice-only calls to Bob's telephone number (TN) will ring only the POTS phone gateway, but calls offering RTT support can be directed automatically to the tablet exclusively or in parallel with the POTS phone.

The presence of the RTT tag in the SIP response allows the network to differentiate between the case where the terminating party supports RTT, but elects not to use it (no transcoder required), and the case where the terminating party doesn't support RTT (transcoder must be provided). In both of these cases, the RTT media line will have a port=0 setting indicating that RTT is not desired, but the presence of the RTT tag in the contact header of the response will allow the network to know that RTT is being declined as a choice rather than because it is not supported.

When transcoding is introduced, the audio and text media streams will be transcoded to G.711 codec using Baudot inband tones along with possible audio. When the IWF introduces transcoding, it will perform an additional offer/answer exchange with the SDP reflecting the IWF media function addressing information. It will also place PCMU (G.711) codec as the preferred codec.

In the call flows that follow:

- WB is used to represent the Wideband AMR codec.
- NB is used to represent the Narrowband AMR codec.
- G711 is used to represent the PCMU codec.
- "p=0" represents port set to zero in the associated SDP media (m) line.
- "text" presents the text media feature tag in the Contact header.
- The IWF is outside of the MGCF.

The flows represent examples; additional and alternative flows are possible.

3.5.1 VoLTE Mobile Origination to PSTN or 3G with VoLTE RTT at Start Of Call

Use Case: Alice is a VoLTE user with an RTT-capable smartphone and calls Bob. Alice knows Bob uses TTY service exclusively, so Alice has her phone configured for RTT operation at the start of the call and as such, Alice's SDP offer includes the m=text line. Bob answers the call using his circuit device with Baudot protocol. The network has provided a transcoder from the start of the call.

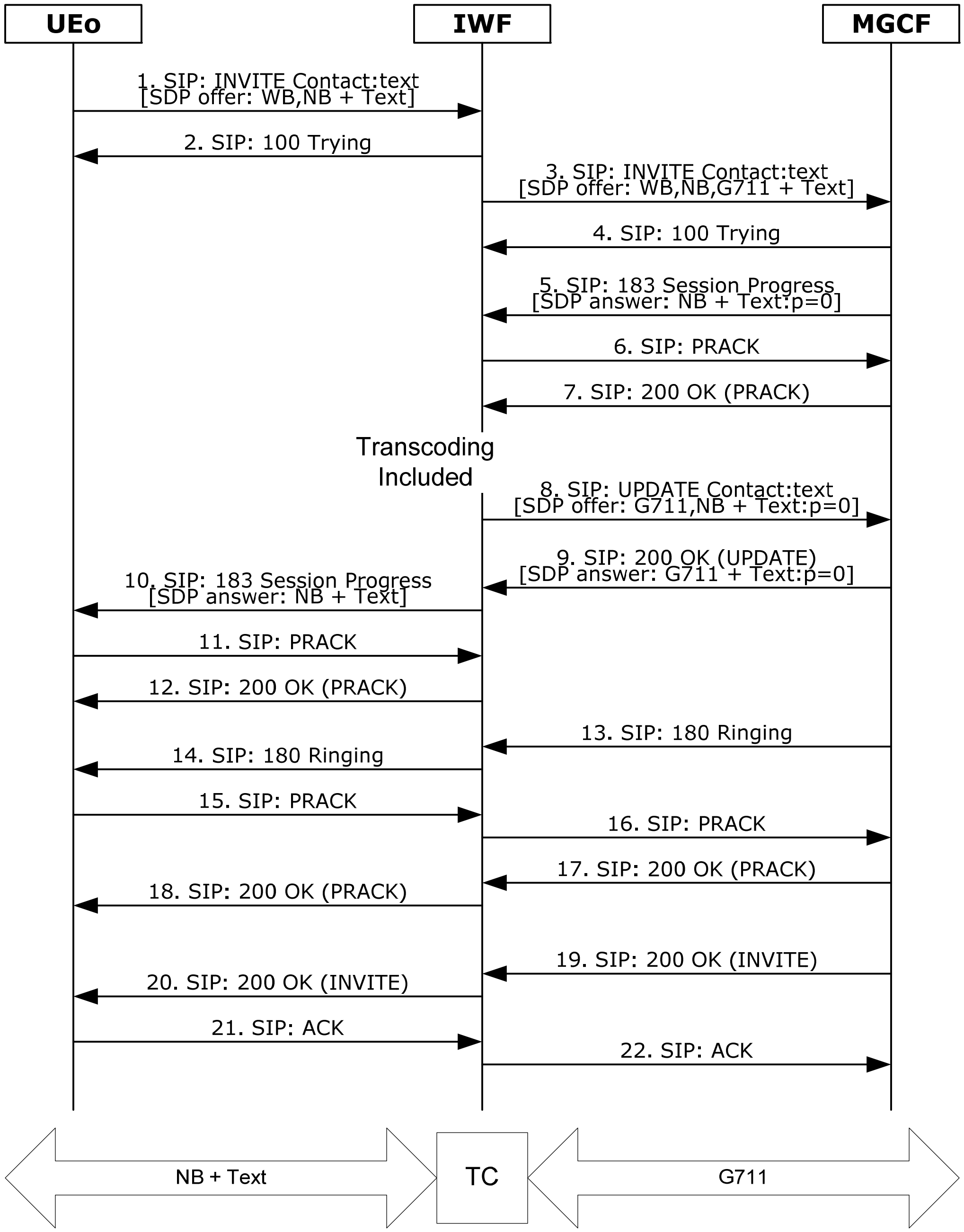

Figure 3.10 – VoLTE Mobile Origination w/RTT to PSTN or 3G

For this flow, the originating UE is configured to use RTT from the start of the call.

Transcoder Invoke:

1. IWF knows that SDP answer without the RTT tag (“text”) to an SDP offer with RTT means that far end is not RTT-capable so RTT-TC is needed. Therefore, for this case, RTT-TC is needed.
2. Reserve RTT-TC.
3. Send UPDATE to far end with RTT-TC port for G711 preferred.
4. After 200 OK (UPD), send SDP answer to UEo with RTT-TC ports supporting NB and RTT.

NOTE: Flow would work the same if G711 was selected in message 5 rather than NB. UPDATE would still be needed to change to TC ports (depends on IWF architecture for managing the various transcoders)

3.5.2 VoLTE Mobile Termination from PSTN or 3G with VoLTE RTT at Start Of Call

Use Case: Alice is using her PSTN line to call Bob's VoLTE phone. Bob uses TTY service exclusively so Bob's VoLTE handset is in TTY mode when he receives Alice's call. Alice knows that Bob uses TTY service exclusively so Alice has placed this call from her Baudot protocol device connected to her PSTN line. Bob answers Alice's call and renegotiates the dialog to include RTT prior to answering the call. The session uses TTY service transcoded by the network from the start of the call.

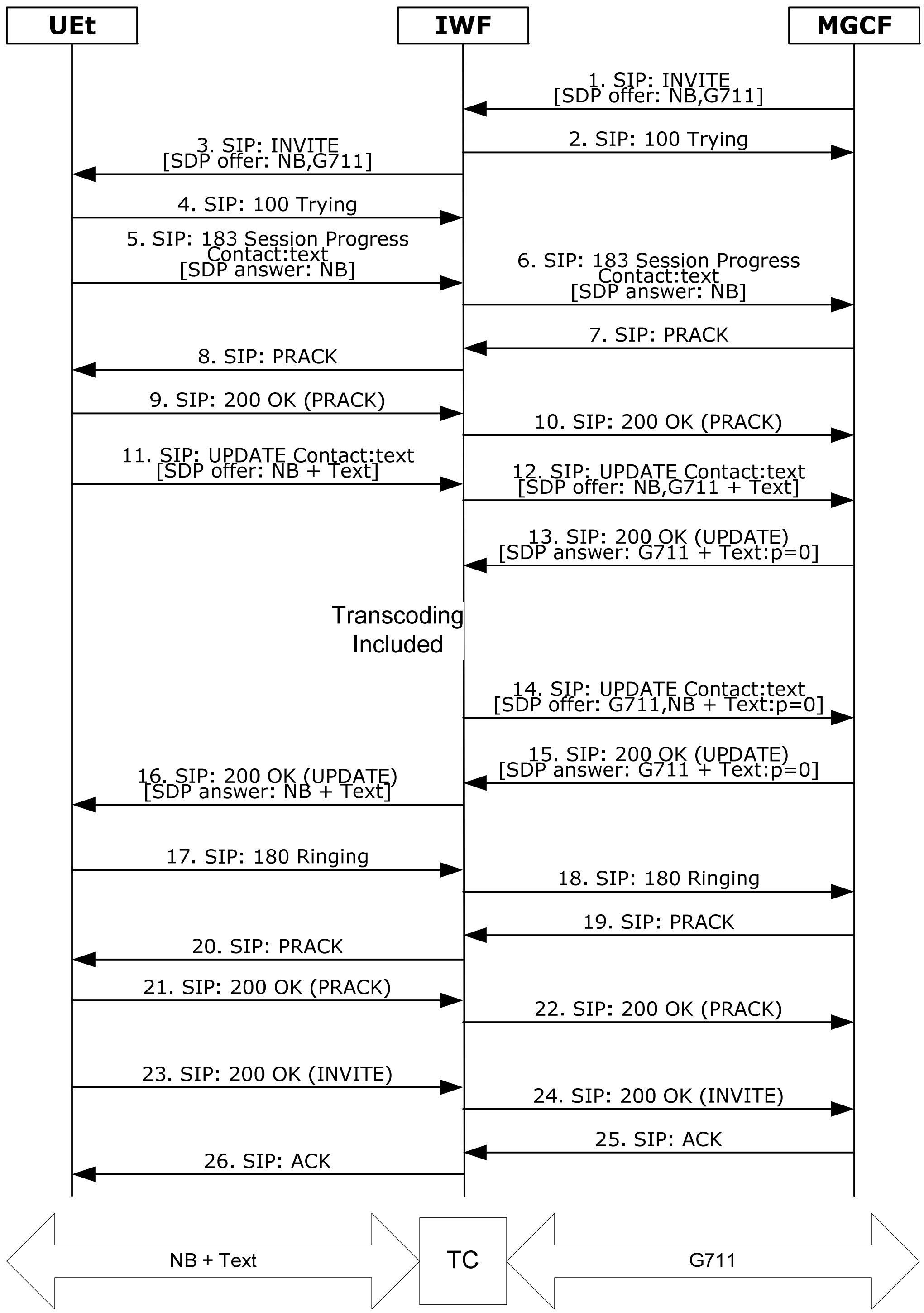

Figure 3.11 – VoLTE Mobile Termination w/RTT From PSTN or 3G

For this flow, the originating UE is configured to use TTY from the start of the call. This will result in the terminating UE immediately sending an UPDATE offering RTT after the completion of the initial offer/answer exchange.

Transcoder Invoke:

1. From step 13, the IWF knows that SDP answer without the RTT tag (“text”) to an offer with RTT means that far end is not RTT-capable so RTT-TC is needed.
2. Reserve RTT-TC.
3. Send UPDATE to far end with RTT-TC port for G711 preferred.
4. After 200 OK (UPD), send SDP answer to UEt with RTT-TC ports supporting NB and RTT.

NOTE: Timing of UPDATE vs. 180 response can be flexible from what is shown here.

NOTE: Message 12 must include m=text line, but whether it should re-offer it or send it as port=0 is unknown; the former is shown here.

3.5.3 VoLTE Mobile Origination with RTT to VoLTE Mobile Termination without RTT

Use Case: Alice is a VoLTE user with an RTT-capable smartphone and calls Carol. Alice doesn’t know Carol but knows she is a friend of Bob and so might be a TTY service user. Alice has configured her phone for TTY operation at the start of the call and as such Alice’s SDP offer includes the m=text line. Carol is also a VoLTE user but is not a TTY service user so while her phone supports RTT, it is not configured to use it and declines the m=text line in Alice’s SDP offer. The call is answered as a voice call and no transcoder is needed nor provided.

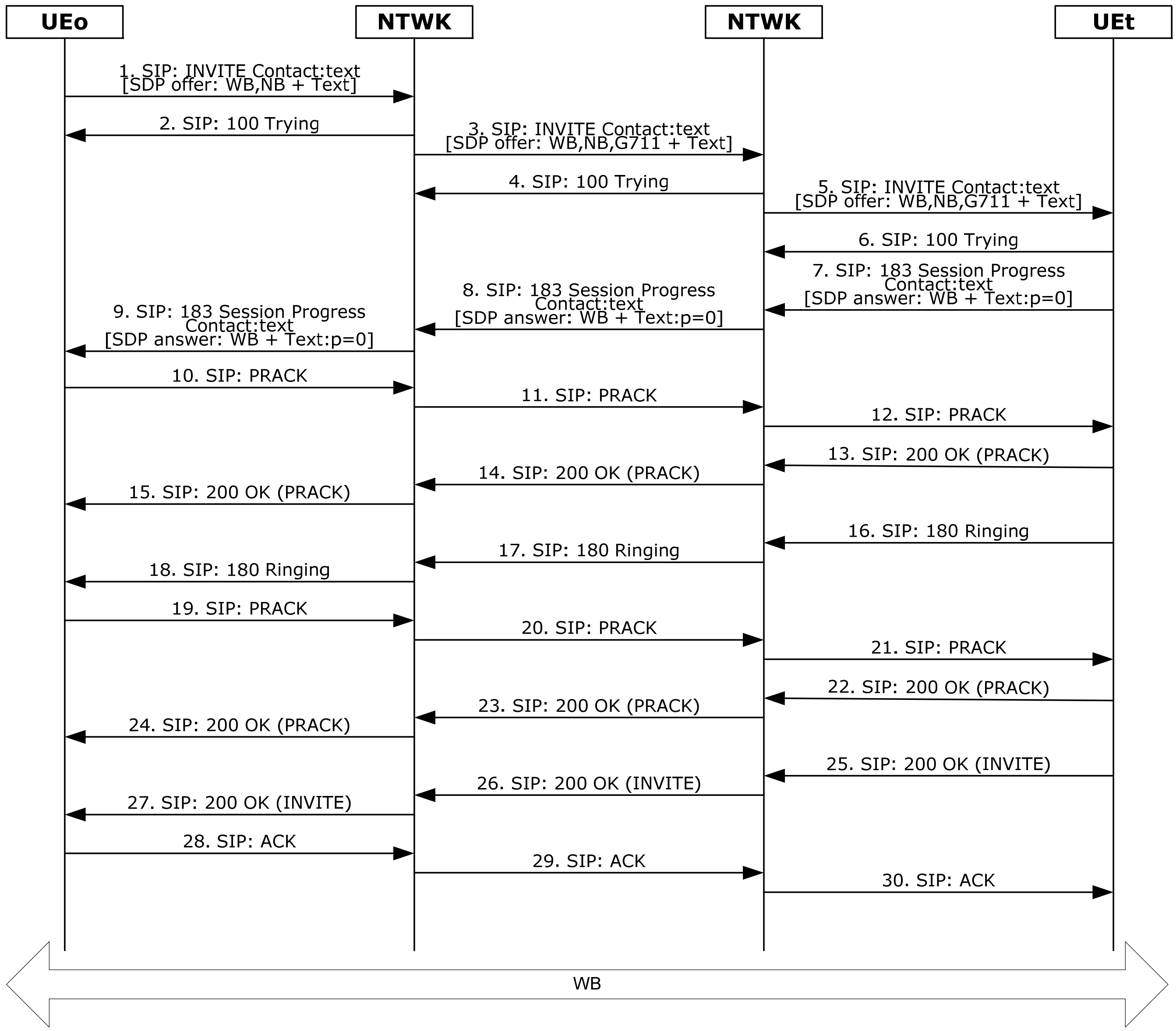

Figure 3.12 – VoLTE Mobile Origination with RTT to VoLTE MT without RTT

For this flow, the originating UE is configured to use TTY service from the start of the call and the terminating UE is configured to ***not*** use TTY/RTT from the start of the call.

> NOTE: Presence of RTT tag ("text") in SDP answer suppresses transcoder logic at IWF. In this case the terminating UE is indicating that it supports RTT but does not want to use it.

3.5.4 VoLTE Mobile Origination with TTY to VoLTE Mobile Termination with RTT

Use Case: Alice is a VoLTE user with an RTT-capable smartphone and calls Bob. Alice knows Bob uses TTY service exclusively so Alice has her phone configured for TTY operation at the start of the call and as such, Alice's SDP offer includes the m=text line. Bob answers the call using his RTT-capable VoLTE smartphone which is also configured for TTY operation at the start of the call. The network sees that RTT is offered and accepted in the SDP answer and so no transcoder is needed.

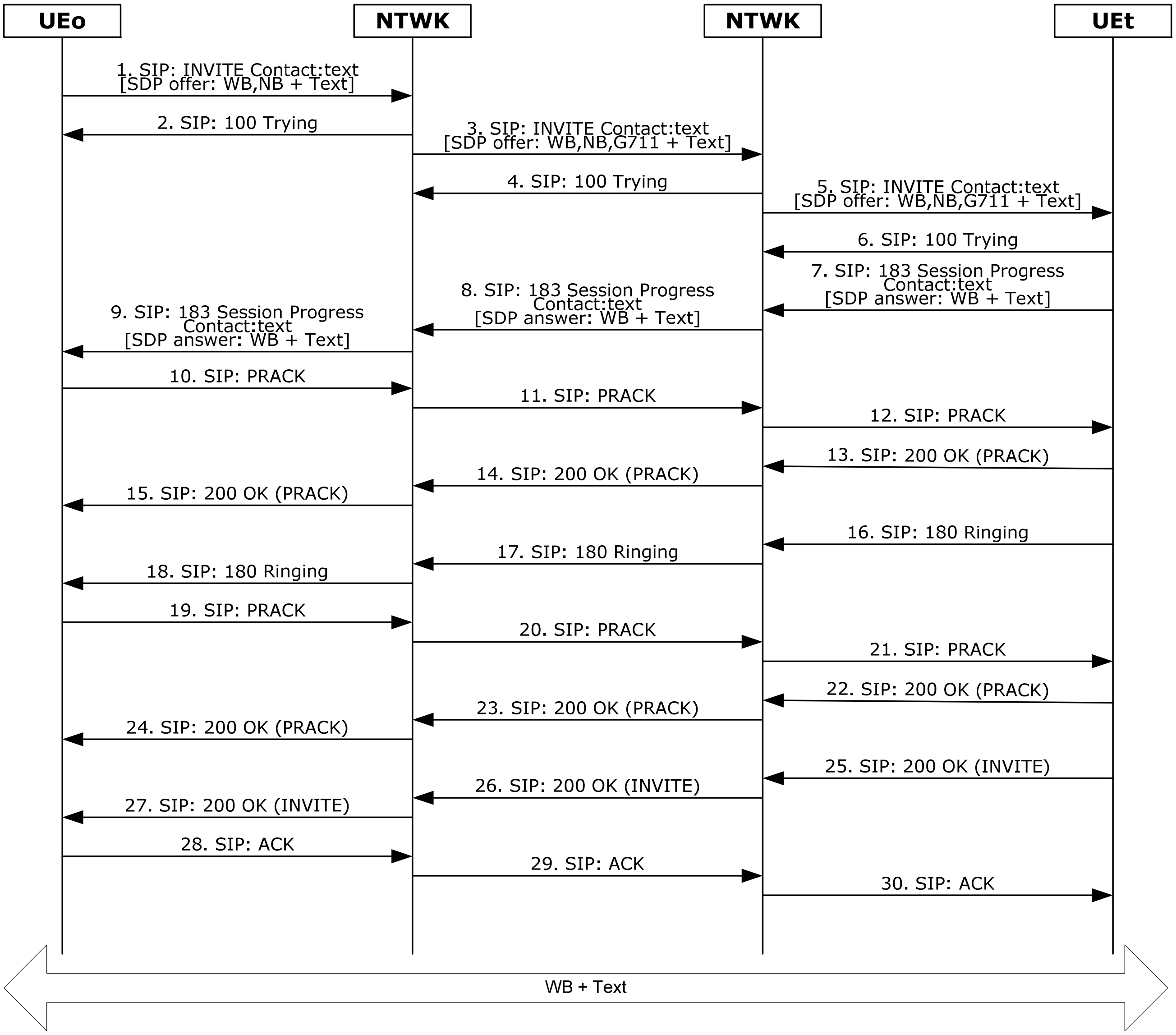

Figure 3.13 – VoLTE Mobile Origination with RTT to VoLTE Mobile Termination with RTT

For this flow, both the originating and terminating UEs are configured to use TTY from the start of the call

NOTE: Presence of RTT tag (“text”) in SDP answer suppresses transcoder logic at IWF.

3.5.5 VoLTE Mobile Origination to PSTN mid-call upgrade to use RTT

Use Case: Alice is a VoLTE user with an RTT-capable smartphone and calls Carol. Alice doesn’t know if Carol is a TTY service user and so has not configured her phone for operation. As such, Alice’s SDP offer does not include the m=text line. Carol is on the PSTN and uses TTY service exclusively. When Carol answers the call, she has her TTY device connected using Baudot protocol. Alice hears the Baudot tones coming from Carol and at step 21 switches her smartphone to RTT mode. The network inserts the transcoder and Alice and Carol can communicate.

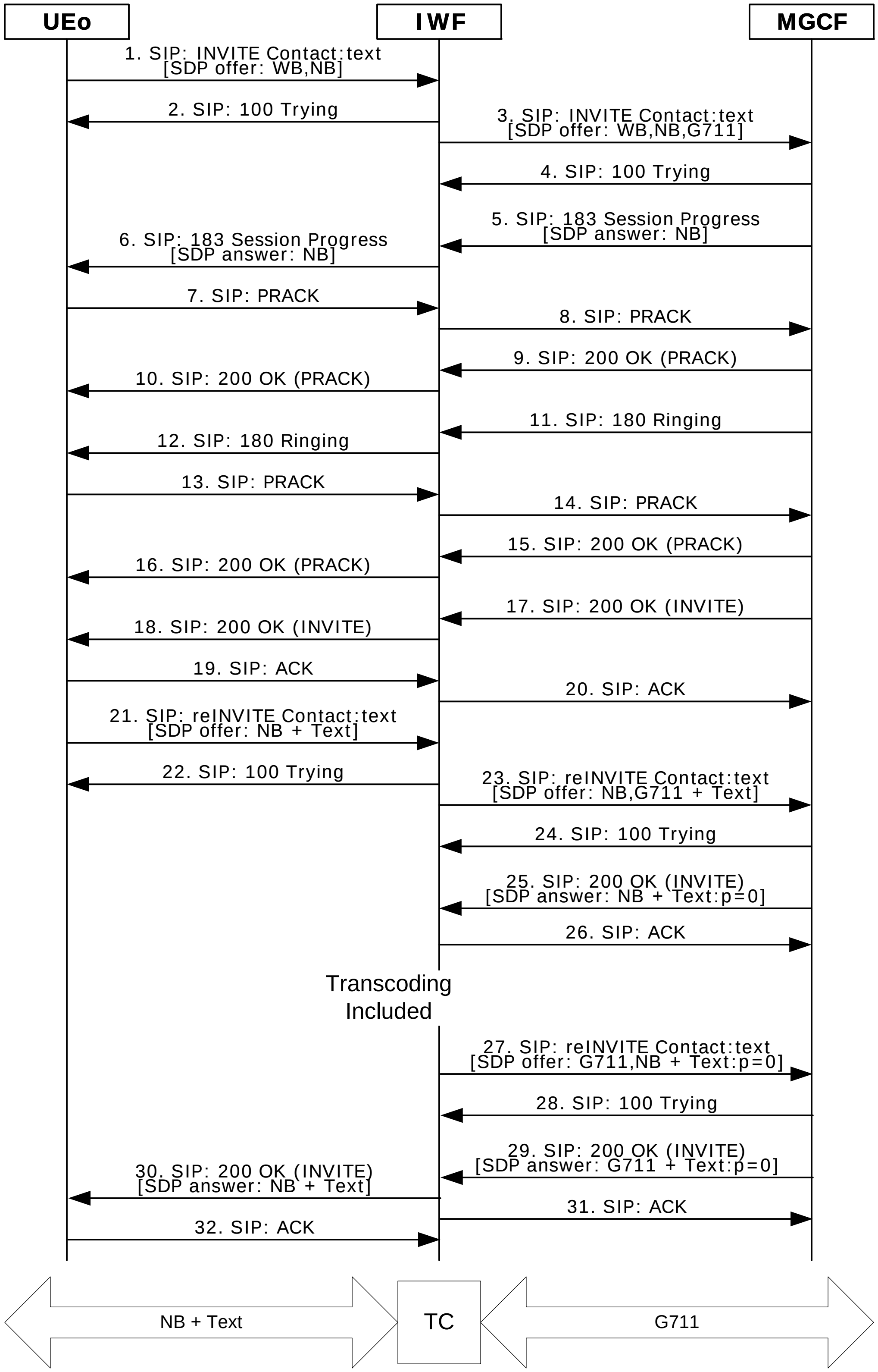

Figure 3.14 – VoLTE Mobile Origination to PSTN Mid-call Upgrade to Use RTT

For this flow, the originating UE is ***not*** configured to use TTY from the start of the call. Instead, the originating user decides to add RTT for TTY service to the call after it has been established.

Transcoder Invoke:

1. From step 25, the IWF knows that SDP answer without the RTT tag (“text”) to an offer with RTT means that far end is not RTT capable so RTT-TC is needed.
2. Reserve RTT-TC.
3. Send reINVITE to far end for G711 preferred with RTT-TC port.
4. After 200 OK (reINV), send SDP answer to UEo with RTT-TC ports supporting NB and RTT.

3.5.6 VoLTE Mobile Termination from PSTN mid-call upgrade to RTT

Use Case: Alice is a VoLTE user with an RTT-capable smartphone. Carol is on the PSTN and uses TTY service exclusively. Carol calls Alice using her TTY device connected to her PSTN line and using Baudot protocol. Alice answers Carol’s calls without knowing that Carol is using TTY service. Alice hears the Baudot tones coming from Carol and at step 21 switches her smartphone to RTT mode. The network inserts the transcoder and Alice and Carol can communicate.

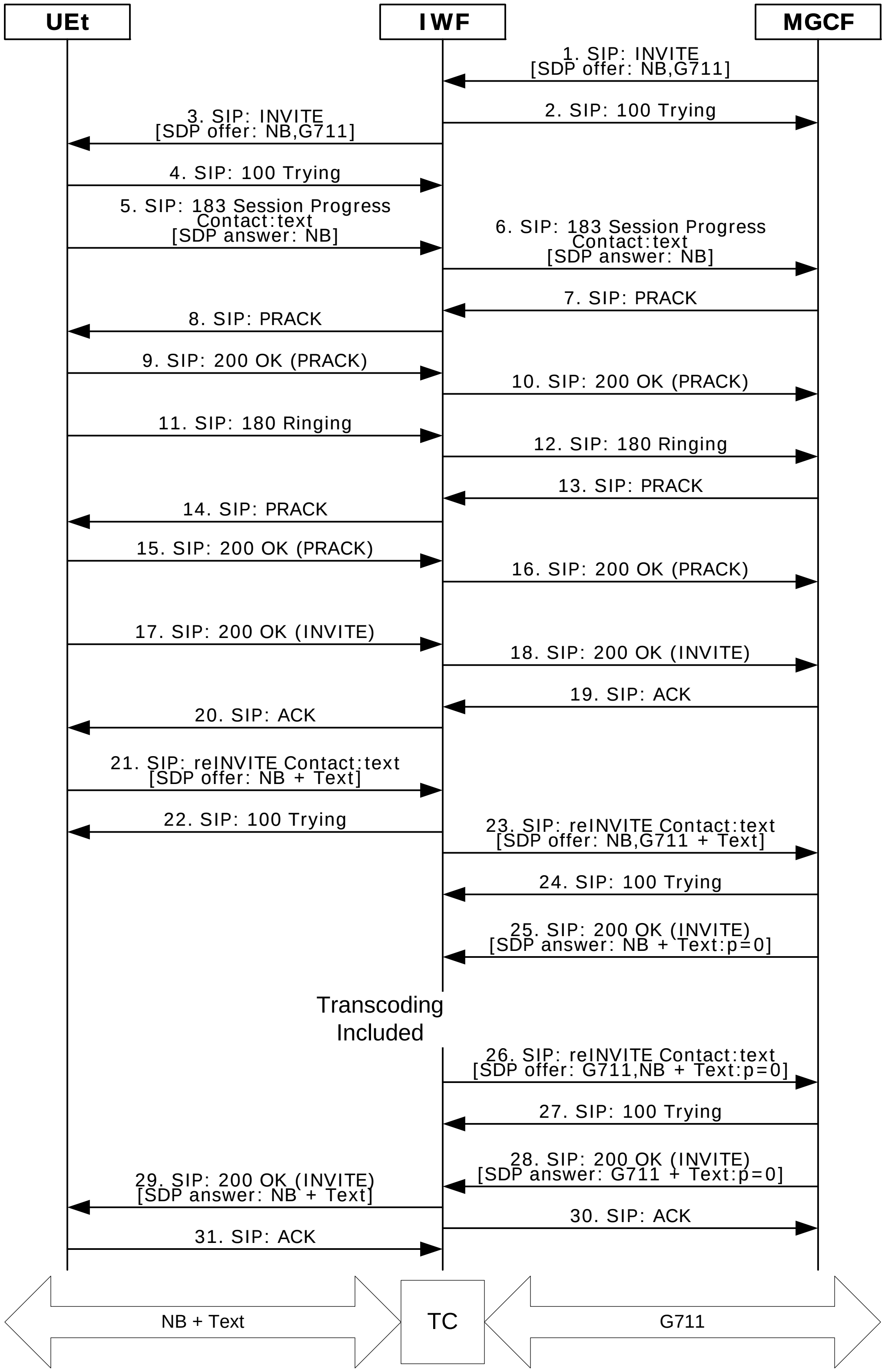

Figure 3.15 – VoLTE Mobile Termination From PSTN Mid-call Upgrade to RTT

For this flow, the terminating UE is *not* configured to use TTY from the start of the call. Instead, the terminating user decides to add RTT to the call after it has been established.

Transcoder Invoke:

1. From step 25, the IWF knows that SDP answer without the RTT tag (“text”) to an offer with RTT means that far end is not RTT-capable so RTT-TC is needed.
2. Reserve RTT-TC.
3. Send reINVITE to far end for G711 preferred with RTT-TC port.
4. After 200 OK (reINV), send SDP answer to UEt with RTT-TC ports supporting NB and RTT.

NOTE: Unclear in step 23 if IWF includes G711 in reINVITE automatically.

3.5.7 VoLTE to VoLTE mid-call upgrade to RTT

Use Case: Alice is a VoLTE user with an RTT-capable smartphone and calls Bob. Alice has her phone configured for RTT operation at the start of the call and as such, Alice’s SDP offer includes the m=text line. Bob answers the call using his RTT capable VoLTE smartphone that is *not* configured for RTT operation at the start of the call, so the call starts as an audio call. Bob then switches his phone to RTT mode and at step 31 the text media type is added to the session. Alice and Bob can communicate by either voice or TTY service.

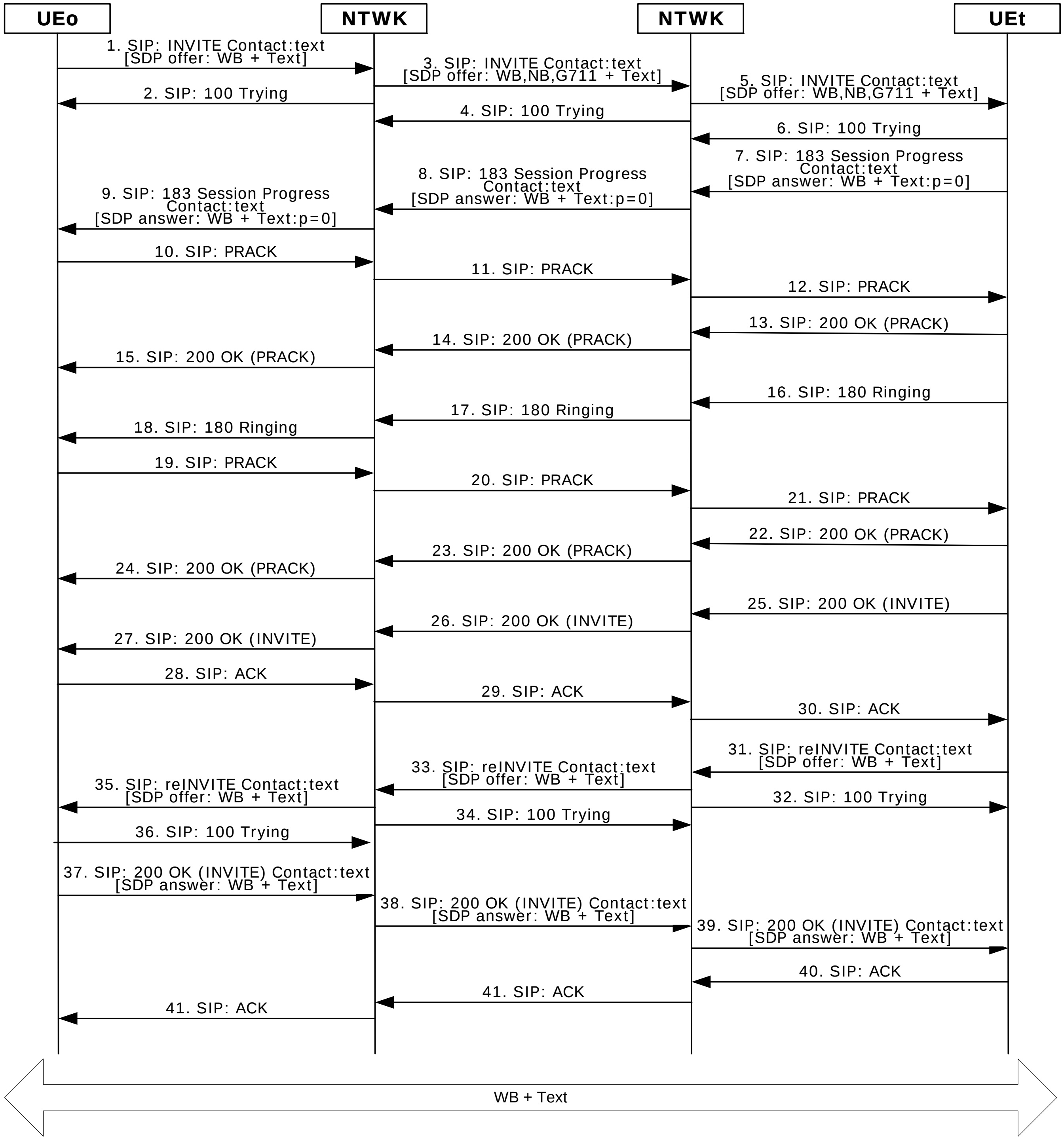

Figure 3.16 – VoLTE to VoLTE with RTT Mid-call Upgrade

For this flow, the originating UE is configured to use TTY from the start of the call and the terminating UE is configured to *not* use TTY from the start of the call. The terminating user decides to add RTT to the call after it has been established.

3.5.8 VoLTE with RTT to VoIP

Use Case: Alice is a VoLTE user with an RTT-capable smartphone and calls Bob. Alice knows Bob uses TTY service exclusively so Alice has her phone configured for RTT operation at the start of the call and as such, Alice's

SDP offer includes the m=text line. Bob uses a VoIP landline service that supports POTS phones via an RJ11 interface. Bob answers the call using his TTY device with Baudot protocol. The network has provided a transcoder from the start of the call so Alice and Bob can communicate.

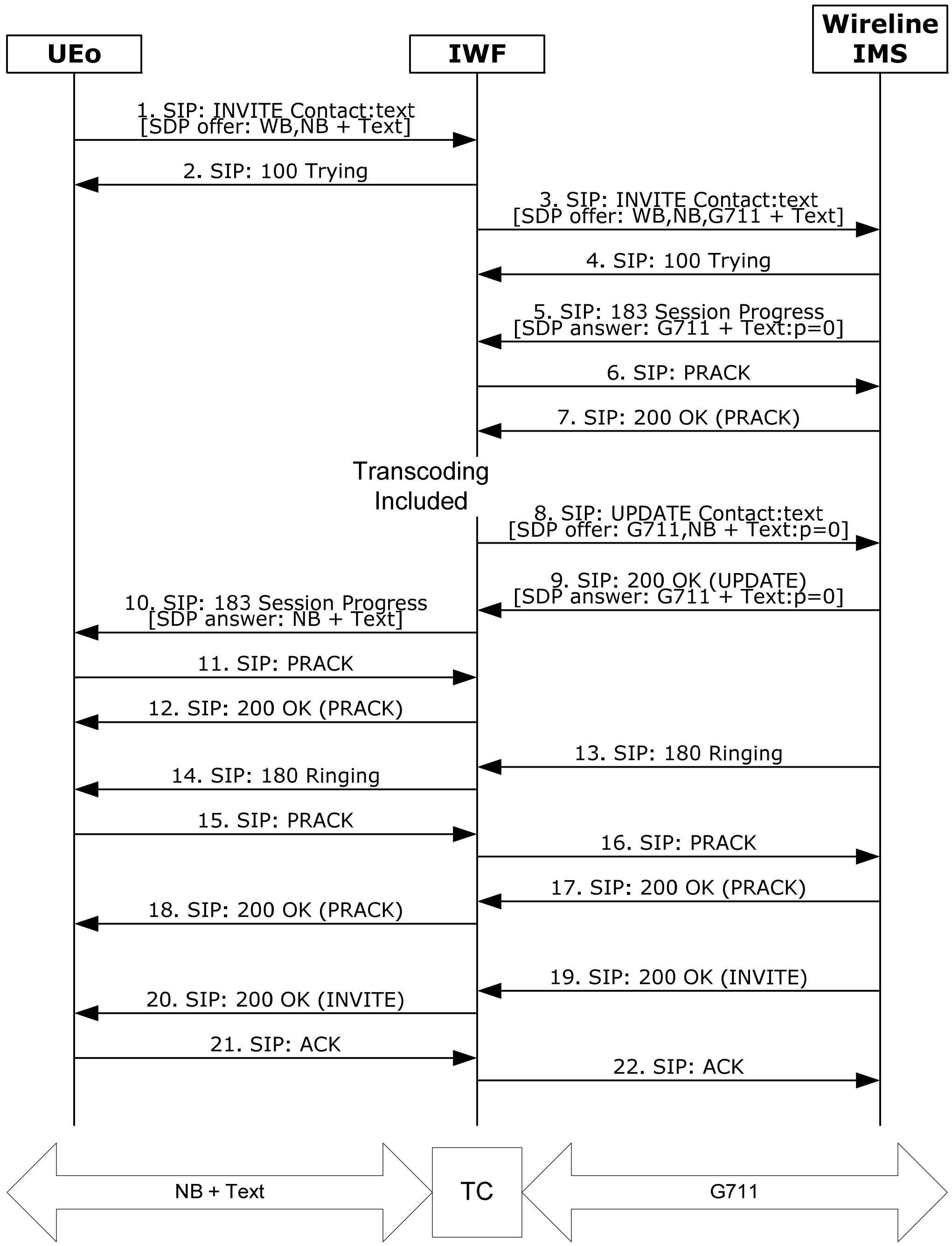

Figure 3.17 – VoLTE Mobile Origination with RTT to VoIP

For this flow, the originating UE is configured to use TTY from the start of the call.

Transcoder Invoke:

1. From step 5, the IWF knows that SDP answer without the RTT tag (“text”) to an offer with RTT means that far end is not RTT-capable so RTT-TC is needed.
2. Reserve RTT-TC.

3. Send UPDATE to far end with RTT-TC port for G711 preferred.
4. After 200 OK (UPD), send SDP answer to UEo with RTT-TC ports supporting NB and RTT.

3.5.9 VoIP to VoLTE with RTT

Use Case: Alice is using her VoIP line with a POTS phone to call Bob’s VoLTE phone. Bob uses TTY service exclusively so Bob’s VoLTE handset is in TTY mode when he receives Alice’s call. Alice knows that Bob uses TTY service exclusively so Alice has placed this call from her Baudot protocol TTY device connected to her PSTN line. Bob answers Alice’s call and session uses TTY service transcoded by the network from the start of the call.

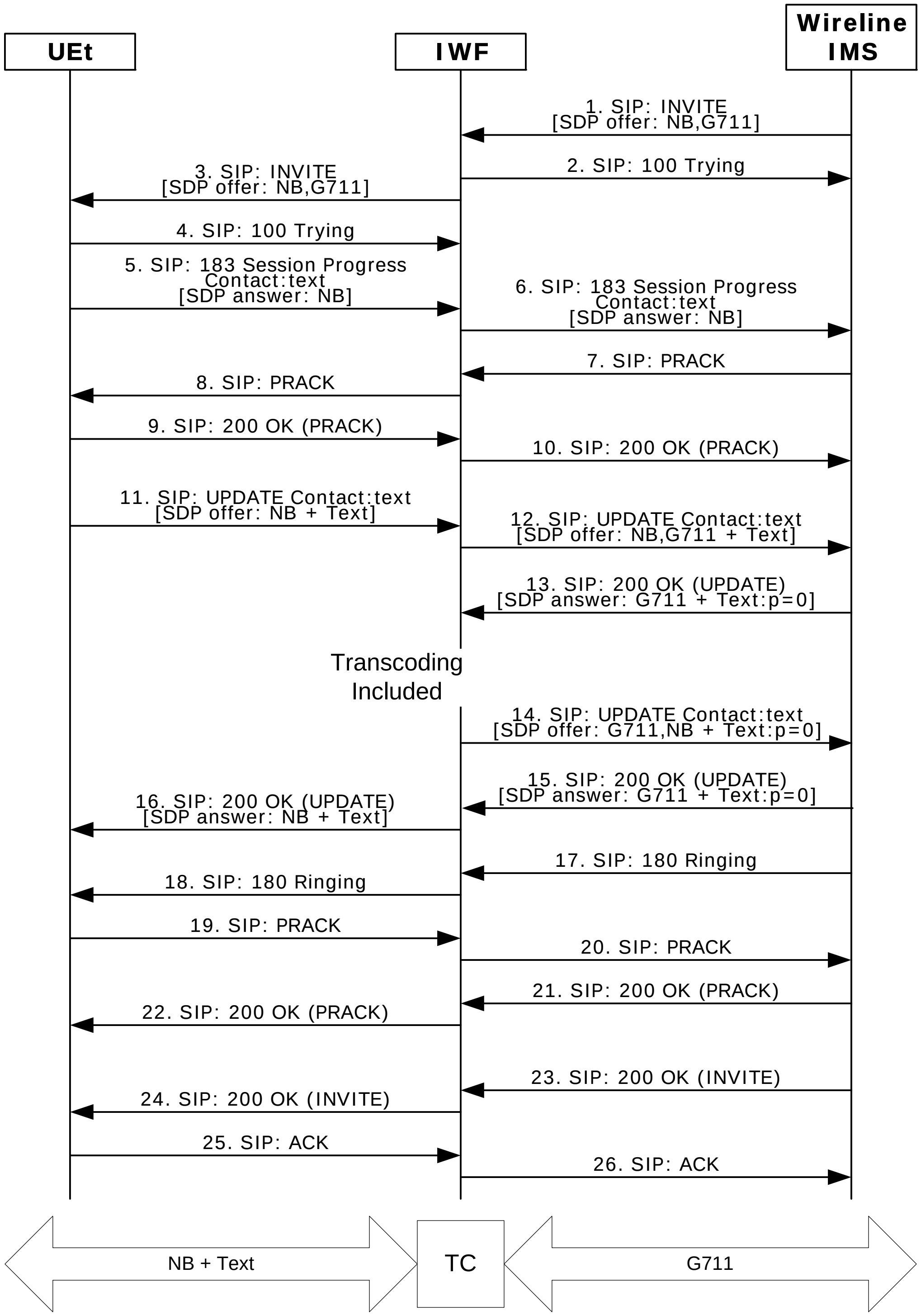

Figure 3.18 – VoIP to VoLTE with RTT

For this flow, the terminating UE is configured to use RTT from the start of the call. As a result, the terminating user will offer RTT immediately after the completion of the initial offer/answer exchange.

Transcoder Invoke:

1. From step 13, the IWF knows that SDP answer without the RTT tag ("text") to an offer with RTT means that far end is not RTT-capable so RTT-TC is needed.
2. Reserve RTT-TC.

3. Send UPDATE toward Wireline IMS device with RTT-TC port for G711 preferred.
4. After 200 OK (UPD), send SDP answer to UEt with RTT-TC ports supporting NB and RTT.

3.6 Media Gateway Conversion Procedures

3.6.1 General

Before text conversation can begin, a call shall first be established between end users. The SIP user may request a text connection from the beginning of a call, or add a request for RTT media at a later stage in a call that was originally established with audio and/or video only.

In addition to the control plane interworking between using SIP and ISUP (when interworking from IP to CS network), the network shall support the negotiation of the RTT payload type (T.140 Text Conversation MIME media type as specified by IETF RFC 4103 [RFC 4103]) in a distinct SDP m-line for text media.

Interworking between RTT and PSTN TTY is provided by introducing conversion in the IWF between IP-based RTT via RTP and modem-based transmission of RTT using ITU-T Recommendation V.18 [V.18] or any of its specific sub-modes.

The conversion function can be seen as a context containing a Text/RTP termination plus a voice/RTP termination and an ITU-T Recommendation V.18 [V.18] text telephony termination with multiplexed V.18 and voice via PCM.

It can be symbolically documented as follows.

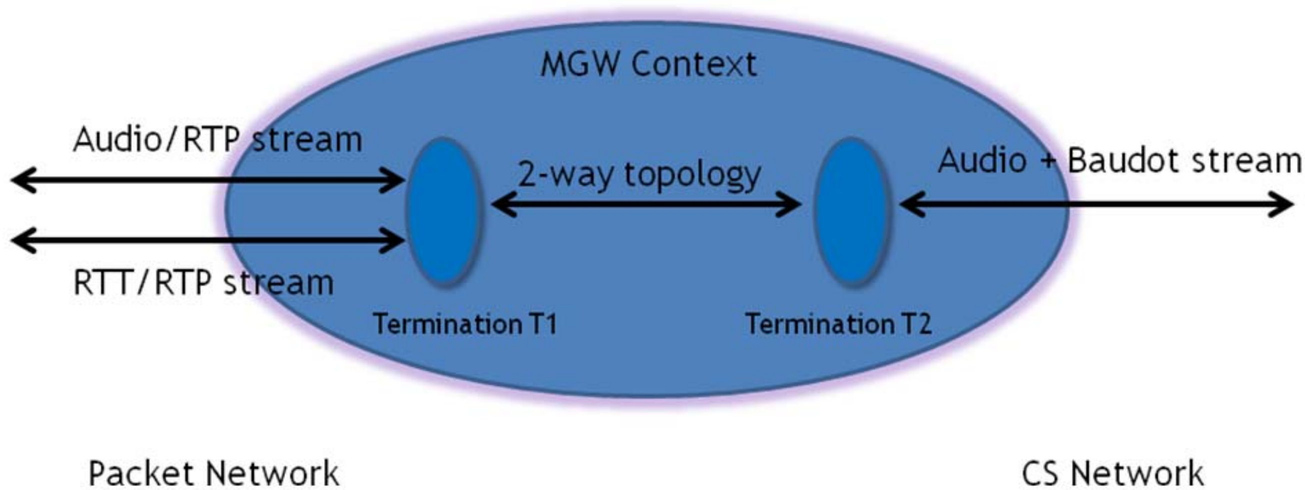

Figure 3.19 – MGW Conversion Model

If the SIP terminal requests RTT telephony, the call shall be set up with the RTT media type in parallel to the voice media, and the IWF/MGW will insert the Interworking function between RTT and V.18. On the contrary, if the SIP terminal does not request RTT support, no Interworking function is necessary (which would represent the majority of calls).

3.7 Interactions with Other Services

3.7.1 Emergency Service Considerations

If an operator implementing GTT elects to offer access to Emergency Services through this feature for a specific host environment, the following must be considered.

If the emergency services only support limited types of text conversation devices, conversion from the users host environment to the one used by the emergency service may be configured.

Other host environment-specific considerations for emergency calls are described in clauses below.

3.7.1.1 Delivery of GTT from IMS Originating Networks to i3 ESInets & Legacy Emergency Services Networks

ATIS-0700015 [ATIS-0700015] describes support for voice and GTT Emergency Sessions originated by IMS subscribers and delivered to i3 ESInets and legacy Emergency Services Networks. The following is an example call flow in which GTT originated by an IMS subscriber is delivered to an i3 ESInet. A user that is served by an IMS originating network that conforms to ATIS-0700015 [ATIS-0700015], and who wishes to immediately offer RTT, can send an initial INVITE that includes the text media type as well as audio. Since the Americans with Disabilities Act (ADA) requires all Public Safety Answering Points (PSAPs) to provide direct, equal access to their services for people with disabilities who use teletypewriters (TTYs), it is assumed that i3 PSAPs (and i3 Legacy PSAP Gateways on behalf of legacy PSAPs) that are served by i3 ESInets will support and accept the offer with RTT, and will return an SDP answer including the text media type and the audio media type.

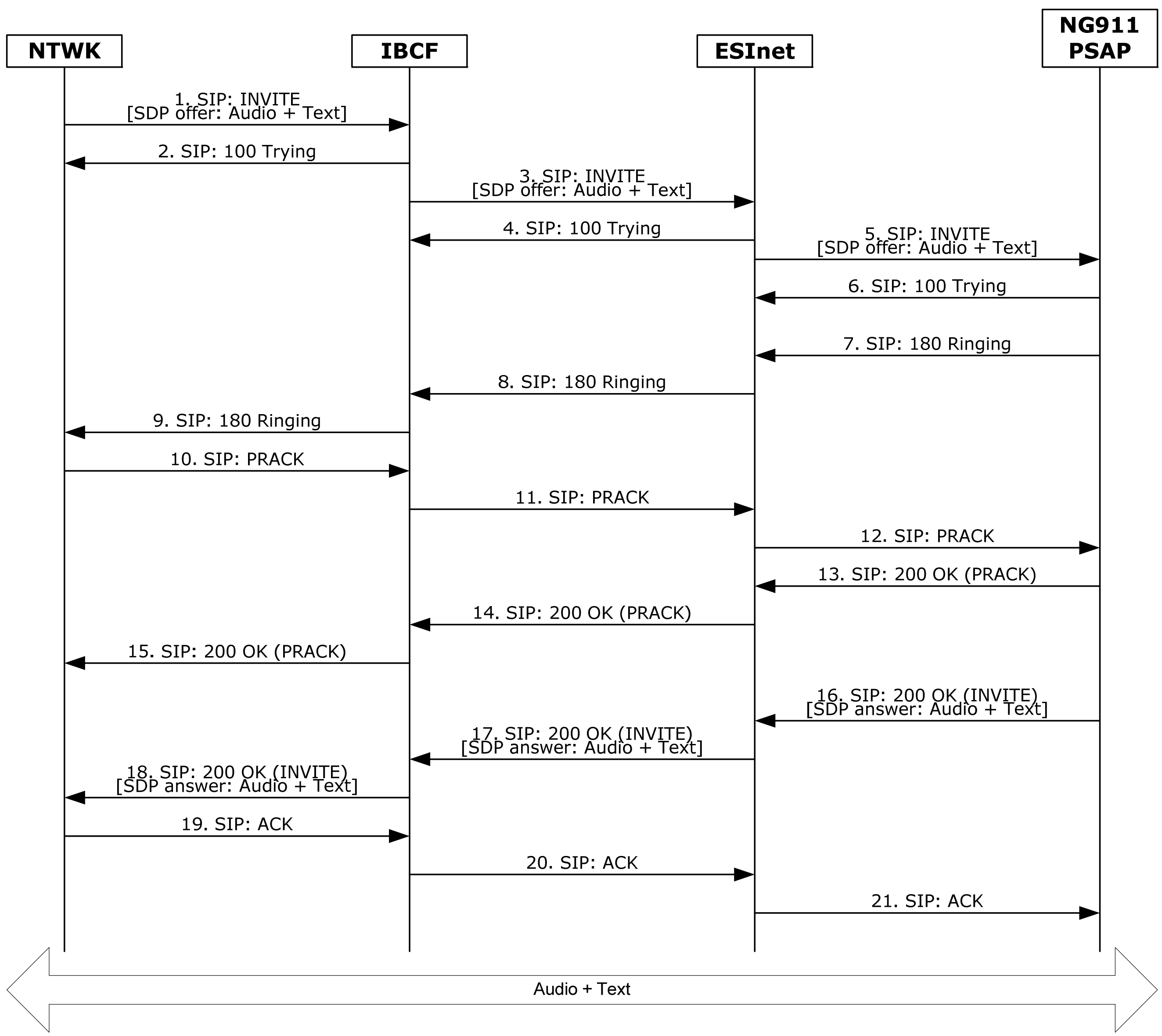

Figure 3.20 – GTT Emergency Call to i3-PSAP

A call flow illustrating the scenario in which GTT originated by an IMS subscriber is delivered to a Selective Router in a legacy Emergency Services Network is provided below.

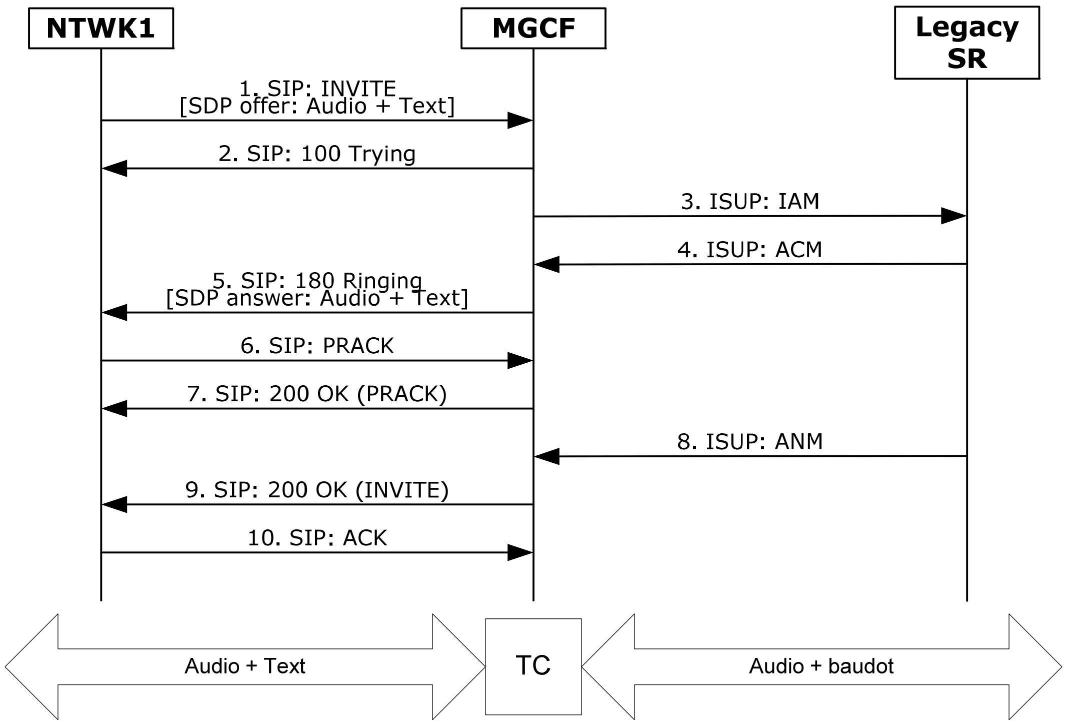

Figure 3.21 – GTT Emergency Call to Legacy Selective Router

3.7.1.2 Delivery of TTY/GTT via an IMS Emergency Services Network

ATIS ESIF defines the application of IMS to Emergency Services Networks within North America. It addresses the interconnection of IMS Emergency Services Networks with legacy (TDM-based), IMS, and non-IMS IP-based originating networks, and the delivery of emergency calls to both i3 and legacy PSAPs.

The following subclauses address call scenarios involving IMS Emergency Services Networks including:

- The delivery of a TTY origination from a legacy network to an i3 PSAP.
- The delivery of a TTY origination from a legacy network to a legacy PSAP.
- The delivery of GTT from an IP-based originating network to an i3 PSAP.
- The delivery of GTT from an IP-based originating network to a legacy PSAP.

3.7.1.2.1 Delivery of Legacy TTY Origination to an i3 PSAP

In the call flow illustrated in Figure 3.22, a user served by a legacy originating network uses TTY communications to originate an emergency call to an i3 PSAP that is served by an IMS Emergency Services Network. In establishing the connection between the user and the PSAP over which Baudot tones will flow, the emergency call flows from the originating network to an i3 Legacy Network Gateway (LNG) outside of the IMS Emergency Services Network that generates a SIP INVITE message with an SDP offer that includes ‘audio’ and ‘text’. The LNG sends the SIP INVITE message to the IMS Emergency Services Network via an IBCF. The IMS Emergency Services Network determines that the session is destined for an i3 PSAP, and delivers the SIP INVITE message (containing an SDP offer that includes ‘audio’ and ‘text’) via an IBCF to the i3 PSAP. The i3 PSAP generates an answer indicating acceptance of ‘audio’ and ‘text’.

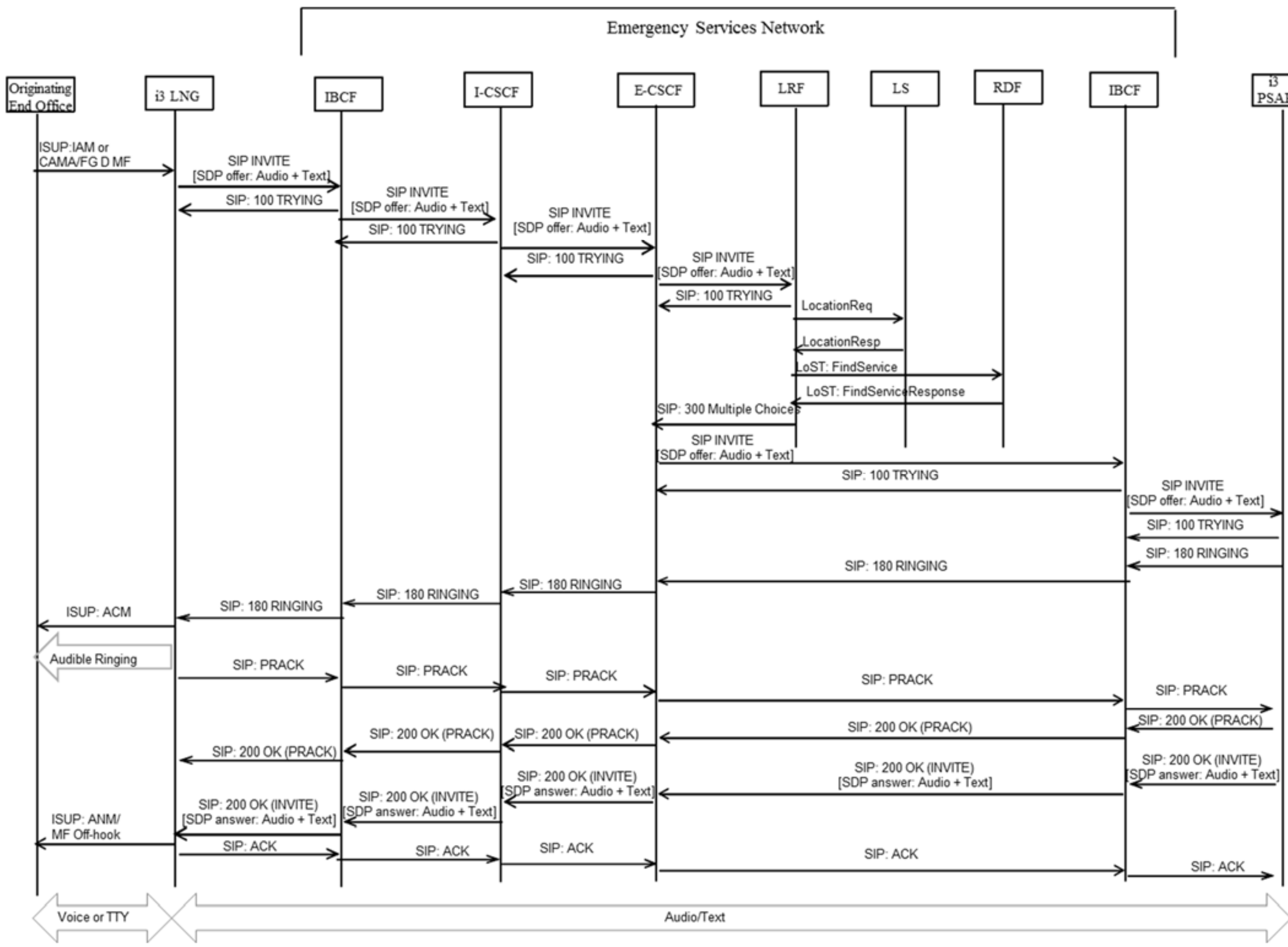

Figure 3.22 – Legacy TTY Origination to i3 PSAP

3.7.1.2.2 Delivery of Legacy TTY Origination to a Legacy PSAP

Figure 3.23 illustrates a call flow in which a user served by a legacy originating network uses TTY communications to originate an emergency call to legacy PSAP that is served by an IMS Emergency Services Network. In establishing the connection between the user and the PSAP over which Baudot tones will flow, the emergency call flows from the originating network to an i3 LNG that generates and sends a SIP INVITE message with an SDP offer that includes ‘audio’ and ‘text’ via an IBCF into the IMS Emergency Services Network. The IMS Emergency Services Network determines that the session is destined for legacy PSAP, and routes the emergency call to the legacy PSAP via a Legacy PSAP Gateway (LPG). The E-CSCF in the IMS Emergency Services Network interconnects with the i3 LPG via an IBCF. The i3 LPG generates Traditional or Enhanced MF signalling (as appropriate for the destination PSAP), and sends it to the PSAP.

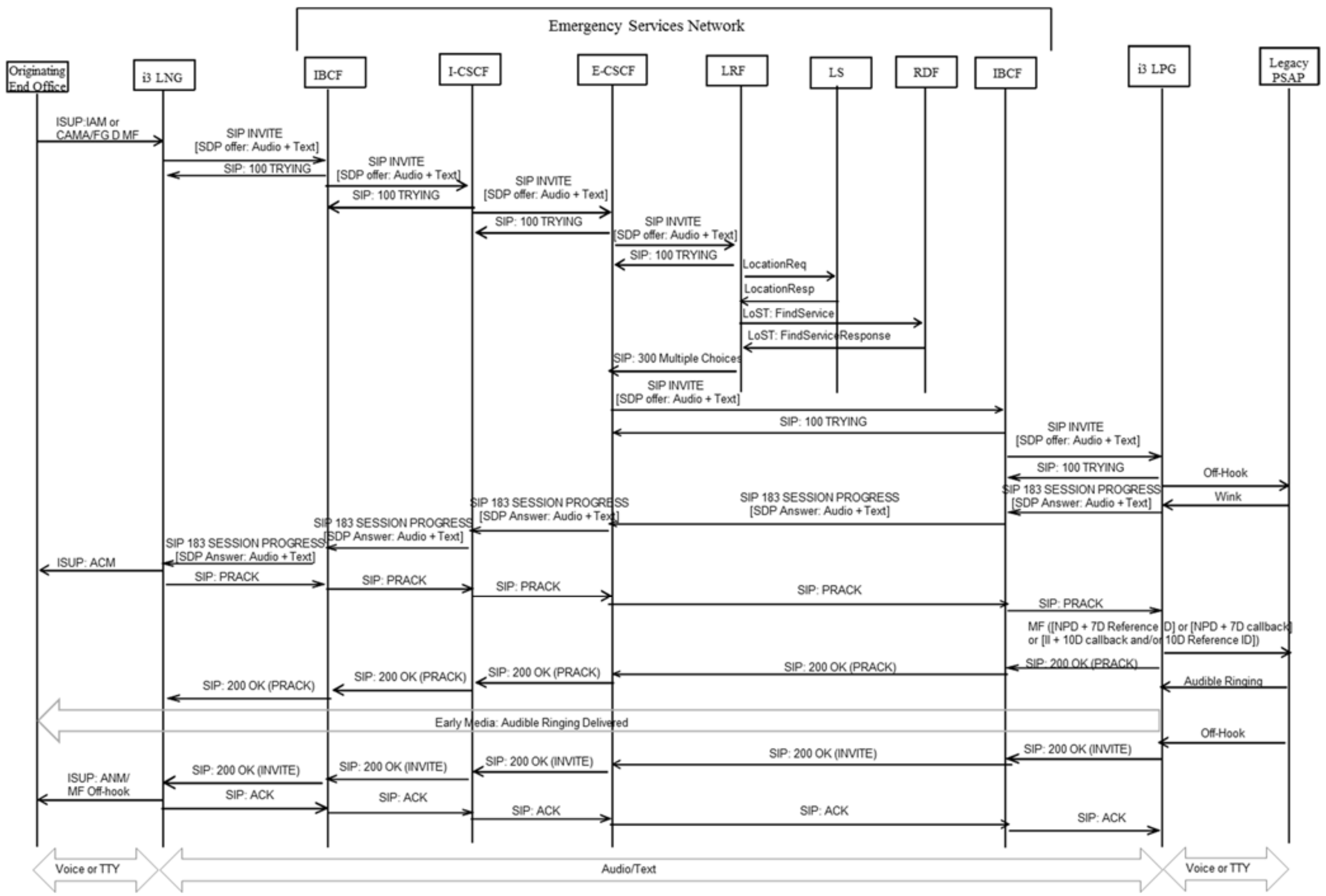

Figure 3.23 – Legacy TTY Origination to Legacy PSAP

3.7.1.2.3 Delivery of GTT Origination to an i3 PSAP

In the call scenario illustrated in Figure 3.24, a user served by an IMS or non-IMS IP-based originating network initiates an emergency session request that includes GTT. A SIP INVITE message containing an SDP offer that includes 'text' and 'audio' is passed from the originating network to an IMS Emergency Services Network via an IBCF. The IMS Emergency Services Network determines that the emergency session request is destined for an i3 PSAP, and sends a SIP INVITE with an SDP offer of 'text' and 'audio' via an IBCF to an i3 PSAP.

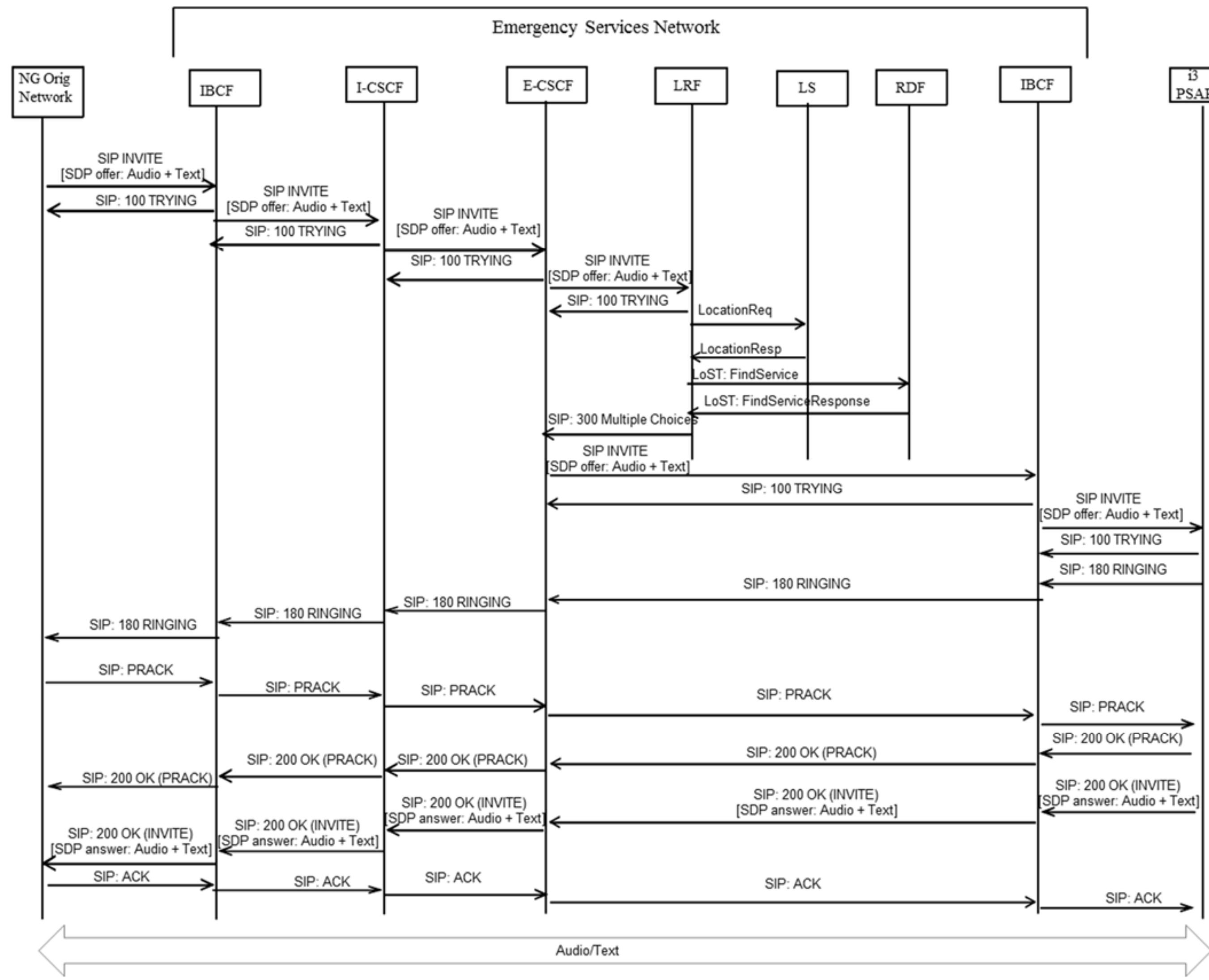

Figure 3.24 – GTT Origination from NG Originating Network to i3 PSAP

3.7.1.2.4 Delivery of GTT Origination to a Legacy PSAP

Figure 3.25 illustrates a call flow in which a user served by an IMS or non-IMS IP-based originating network initiates an emergency session request that includes GTT. A SIP INVITE message containing an SDP offer that includes ‘text’ and ‘audio’ is passed from the originating network to an IMS Emergency Services Network via an IBCF. The IMS Emergency Services Network determines that the emergency session request is destined for a legacy PSAP, and sends a SIP INVITE with an SDP offer of ‘text’ and ‘audio’ to an i3 LPG. The E-CSCF in the IMS Emergency Services Network interconnects with the i3 LPG via an IBCF. The i3 LPG generates Traditional or Enhanced MF signalling (as appropriate for the destination PSAP), and sends it to the legacy PSAP.

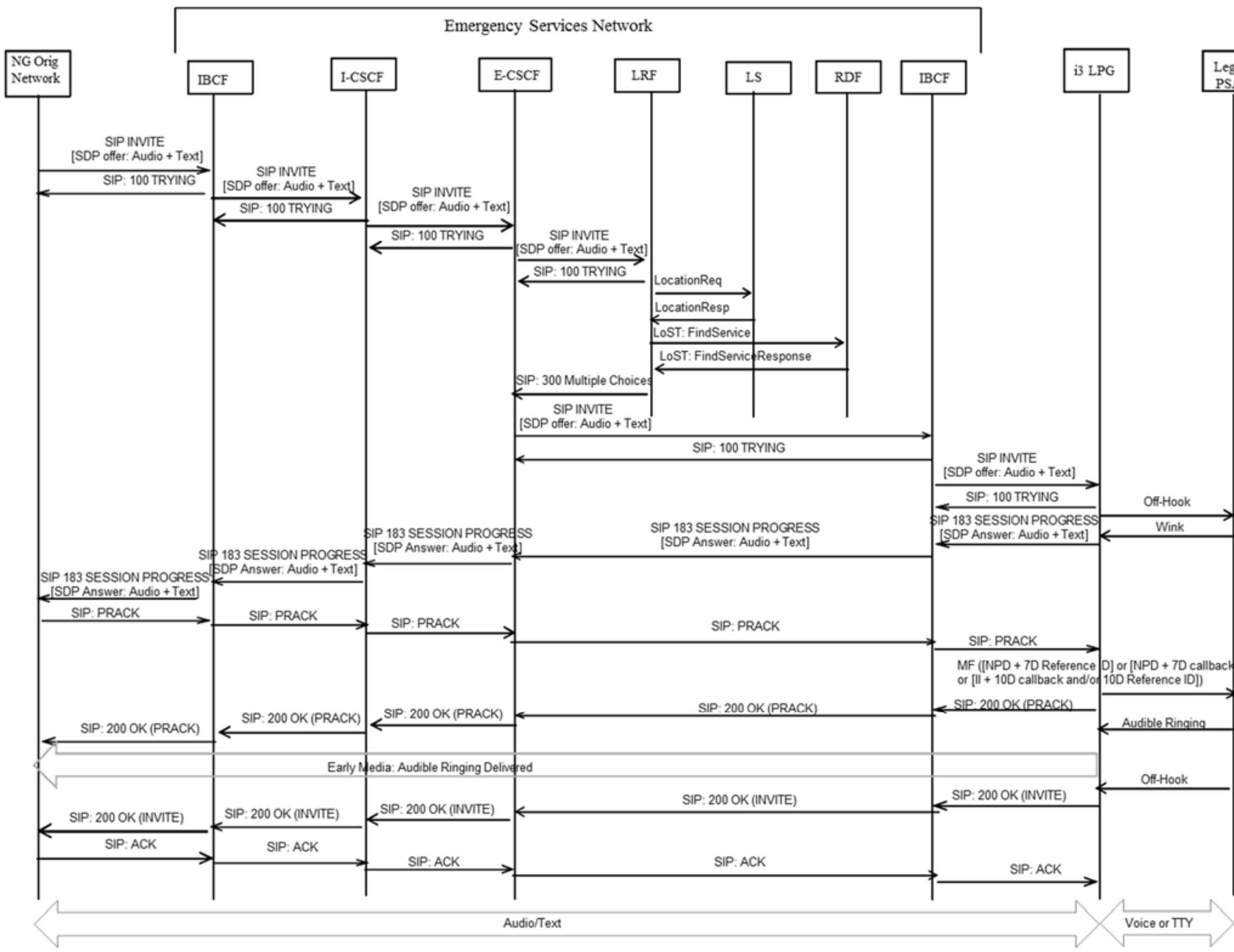

Figure 3.25 – GTT Origination from NG Originating Network to Legacy PSAP

3.7.2 Multi-party Services

Multiplexing audio and RTT media is out of scope for this document.

4 References

4.1 ATIS References

[ATIS-0700015] *ATIS Standard for Implementation of 3GPP Common IMS Emergency Procedures for IMS Origination and ESInet/Legacy Selective Router Termination*, 05-2015.[1]

4.2 3GPP References[2]

[TS 26.226]: *Cellular text telephone modem, General description*, v12.0.0, 09-2014.

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[1] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS) at: < https://www.atis.org/docstore/product.aspx?id=28140 >.

[2] This document is available from the Third Generation Partnership Project (3GPP) at: < http://www.3gpp.org/specs/specs.htm >.

4.3 ITU-T References[3]

[V.18]: *Operational and interworking requirements for DCEs operating in the text telephone mode*, 11-2000.

[T.140]: *Protocol for multimedia application text conversation*, 02-1998; T.140 Addendum 1, 02-2000.

4.4 IETF References[4]

[RFC 2198]: *RTP Payload for Redundant Audio Data*, 09-1997.

[RFC 3840]: *Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)*, 08-2004.

[RFC 4103]: *RTP Payload for Text Conversation*, 06-2005.

[3] These documents are available from the International Telecommunications Union at: < http://www.itu.int/ITU-T/ >.

[4] These documents are available from the Internet Engineering Task Force (IETF) at: < http://www.ietf.org >.