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Packet Switched Voice (over IP) and Video Telephony Services End-to-end System Design Technical Report

FOREWORD

(This foreword is not part of this document)

This document was prepared by 3GPP2 TSG-X.

This document is a new specification.

Revision History

Revision	Date
Version 1.0 Initial Publication	November, 2005

CONTENTS

LIST OF FIGURES

1 Scope

This document describes the end-to-end protocols and procedures for support of Packet Switched Video Telephony (PSVT) and Voice-Over-IP (VoIP) Services over cdma2000®[1] networks. The document describes the end-to-end PSVT and VoIP services and architectures.

PSVT provides support of one-to-one conversational video services between a mobile station and another mobile station or a video terminal on the Internet. VoIP provides support of one-to-one conversational speech services between a mobile station and another mobile station or a voice terminal on the Internet. The designs are compliant with the procedures specified in the 3GPP2 standards [6][7][8][9][10][11][12][13][16][17][19].

2 Introduction

This technical report is an informative document that identifies and summarizes the relevant procedures in other 3GPP2 specifications that are used to provide 3GPP2 PSVT and VoIP services. The language used in this document is informative and is not meant to add or modify procedures already specified in other 3GPP2 standards.

[1] "cdma2000® is the trademark for the technical nomenclature for certain specifications and standards of the Organizational Partners (OPs) of 3GPP2. Geographically (and as of the date of publication), cdma2000® is a registered trademark of the Telecommunications Industry Association (TIA-USA) in the United States."

3 Glossary and Definitions

3.1 Acronyms

1X	cdma2000 1X Air Interface
AAA	Authentication, Authorization, and Accounting
AIR	Accounting Information Record
AES	Advanced Encryption Standard
BSC	Base Station Controller
DSCP	Diff-serve Code Point
H-AAA	Home AAA
HRPD	High Rate Packet Data
HTTP	Hyperlink Text Transfer Protocol
IMS	IP Multimedia Subsystem
IP	Internet Protocol
MS	Mobile Station
NAI	Network Address Identifier
NID	Network Identifier
OTA	Over The Air
PCF	Packet Control Function
PDSN	Packet Data Serving Node
PSVT	Packet Switched Video Telephony
PZID	Packet Zone Identifier
RADIUS	Remote Authentication Dial In User Service
RAN	Radio Access Network
ROHC	Robust Header Compression
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
SBBC	Service Based Bearer Control
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SRTP	Secure Real-time Transport Protocol
UDP	User Datagram Protocol
UIM	User Identity Module
UTC	Universal Coordinated Time
VOIP	Voice over IP
VSA	Vendor Specific Attribute

3.2 Definitions

1X Air interface specification as defined in [6][7][8][9][10].

HRPD Air interface specification as defined in [13].

Media Traffic The RTP/UDP/IP packets transporting the encoded audio, video, or VoIP traffic.

4 References

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

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP2 document, a non-specific reference implicitly refers to the latest version of that document in the same Release as the present document.

[1] **3GPP2:** A.S0008-A, Interoperability Specification (IOS) for High Rate Packet Data (HRPD) Access Network Interfaces - Rev A, March 2006.

[2] **3GPP2:** A.S0009-A, Interoperability Specification (IOS) for High Rate Packet Data (HRPD) Access Network Interfaces - Rev A, March 2006.

[3] **3GPP2:** A.S0013-C, Interoperability Specification (IOS) for cdma2000 Access Network Interfaces – Part 3 Features, February 2005.

[4] **3GPP2:** A.S0014-C, Interoperability Specification (IOS) for cdma2000 Access Network Interfaces – Part 4 (A1, A1p, A2, and A5 Interfaces), February 2005.

[5] **3GPP2:** C.R1001-E v1.0, Administration of Parameter Value Assignments for cdma2000 Standards for Spread Spectrum Systems, October 2005.

[6] **3GPP2:** C.S0001-A v5.0, Introduction for cdma2000 Standards for Spread Spectrum Systems, July 2001.

[7] **3GPP2:** C.S0002-A v6.0, Physical Layer Standard for cdma2000 Standards for Spread Spectrum Systems, February 2002.

[8] **3GPP2:** C.S0003-A v6.0, Medium Access Control (MAC) Standard for cdma2000 Standards for Spread Spectrum Systems, February 2002.

[9] **3GPP2:** C.S0004-A v6.0, Signalling Link Access Control (LAC) Standard for cdma2000 Standards for Spread Spectrum Systems, February 2002.

[10] **3GPP2:** C.S0005-A v6.0, Upper Layer (Layer 3) Signalling Standard for cdma2000 Standards for Spread Spectrum Systems, February 2002.

[11] **3GPP2:** C.S0017-010-A v1.0, Data Service Options for Spread Spectrum Systems: Radio Link Protocol Type 3, July 2004.

[12] **3GPP2:** C.S0017-012-A v1.0, Data Service Options for Spread Spectrum Systems: Service Options 33 and 66, July 2004.

[13] **3GPP2:** C.S0024-A v1.0, cdma2000 High Rate Packet Data Air Interface Specification, April 2004.

[14] **3GPP2:** C.S0039-0 v1.0, Enhanced Subscriber Privacy for cdma2000 High Rate Packet Data, September 2002.

[15] **3GPP2:** C.S0047-0 v1.0, Link-Layer Assisted Service Options for Voice-Over-IP: Header Removal (SO 60) and Robust Header Compression (SO 61), April 2003.

[16] **3GPP2:** S.R0086-A, IMS Security Framework, June 2004.

[17] **3GPP2:** X.S0011, cdma2000 Wireless IP Network Standard.

[18] **3GPP2:** X.S0013-000-0 v2.0, All-IP Core Network Multimedia Domain: Overview, August 2005.

[19] **3GPP2:** X.S0013-002-0 v1.0, All-IP Core Network Multimedia Domain: IP Multimedia Subsystem – Stage 2, February 2004.

[20] **3GPP2:** X.S0013-003-0 v2.0, All-IP Core Network Multimedia Domain: IP Multimedia Session Handling; IP Multimedia Call Model – Stage 2, August 2005.

[21] **3GPP2:** X.S0013-007-0 v1.0, All-IP Core Network Multimedia Domain: IP Multimedia Subsystem – Charging Architecture, February 2004.

[22] **3GPP2:** X.S0013-008-0 v2.0, All-IP Core Network Multimedia Domain: IP Multimedia Subsystem – Accounting Information Flows and Protocol, August 2005.

[23] **IETF:** RFC 3095, Borman, et al, 'RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed', July 2001.

[24] **IETF:** RFC 3261, J. Rosenberg et al, 'SIP: Session Initiation Protocol', June 2002.

[25] **IETF:** RFC 3545, T. Koren, et al. 'Enhanced Compressed RTP (CRTP) for Links with High Delay, Packet Loss and Reordering', July 2003.

5 Protocol Reference Model

5.1 Network Reference Model

The network reference model for a mobile-to-mobile PSVT or VoIP call is illustrated in Figure 1.

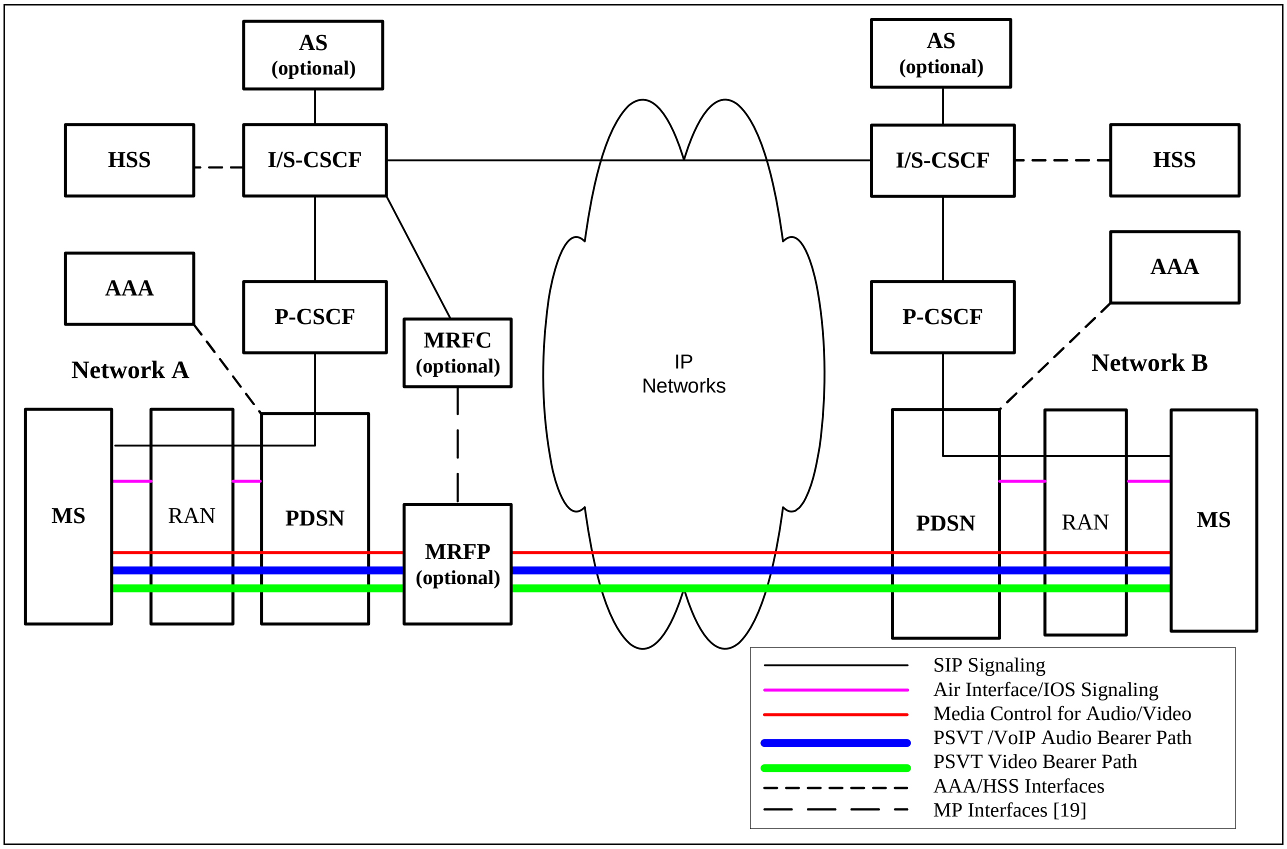

Figure 1 Network Reference Model for Non-roaming Mobile-to-Mobile PSVT or VoIP Call

The function of each entity is listed below:

AAA

The AAA authenticates the subscriber to the access network and sends the list of the subscriber's authorized Flow Profile IDs to the RAN via the PDSN, including the Flow Profile IDs for PSVT or VoIP. The AAA also performs the accounting for the access network.

AS

The Application Server can be a SIP application server (e.g., video mail or voice mail server, etc…) or an OSA gateway as specified in [19].

HSS

The HSS provides the authentication vector to the S-CSCF for IMS authentication. The HSS also performs authorization and accounting for the IMS.

I-CSCF

Interrogating-CSCF (I-CSCF) is the entry point within an operator's network for all session setup attempts destined to a user of that network operator, or a roaming user currently located within that network operator's service area.

P-CSCF

The Proxy-CSCF is the first entry point within the IP Multimedia Subsystem [19]. The P-CSCF behaves like a Proxy, i.e. it accepts requests and services them internally or forwards them on to the MS, an I-CSCF, or the S-CSCF.

PDSN

The PDSN communicates with the MS using service connections for packet data session establishment, to add and remove IP flows, etc. as described in [17]. The PDSN function acts as the first-hop router for IP traffic to and from the MS.

MRFC

The Media Resource Function Controller, in conjunction with the MRFP, provides a set of resources with the MMD core network that are useful in supporting services to subscribers. In the network reference model the MRFC interfaces to the S-CSCF to determine media conversion requirements and instructs the MRFP to perform any necessary conversion of the media stream.

MRFP

The Media Resource Function Processor converts the media formats sent between the mobile stations when the mobile stations use incompatible codecs.

MS

The mobile station contains the video and audio codecs that provide the multimedia interface to the user. The mobile station also contains the SIP user agent that communicates with the other terminal (MS or general IP video or voice telephony device) through the core network CSCFs.

RAN

The Radio Access Network communicates with the MS using the service option [12][15] or link flow [13] to transport packet data over the radio link.

S-CSCF

The Serving-CSCF (S-CSCF) performs the session control services for the MS. It maintains a session state as needed by the network operator for support of the services.

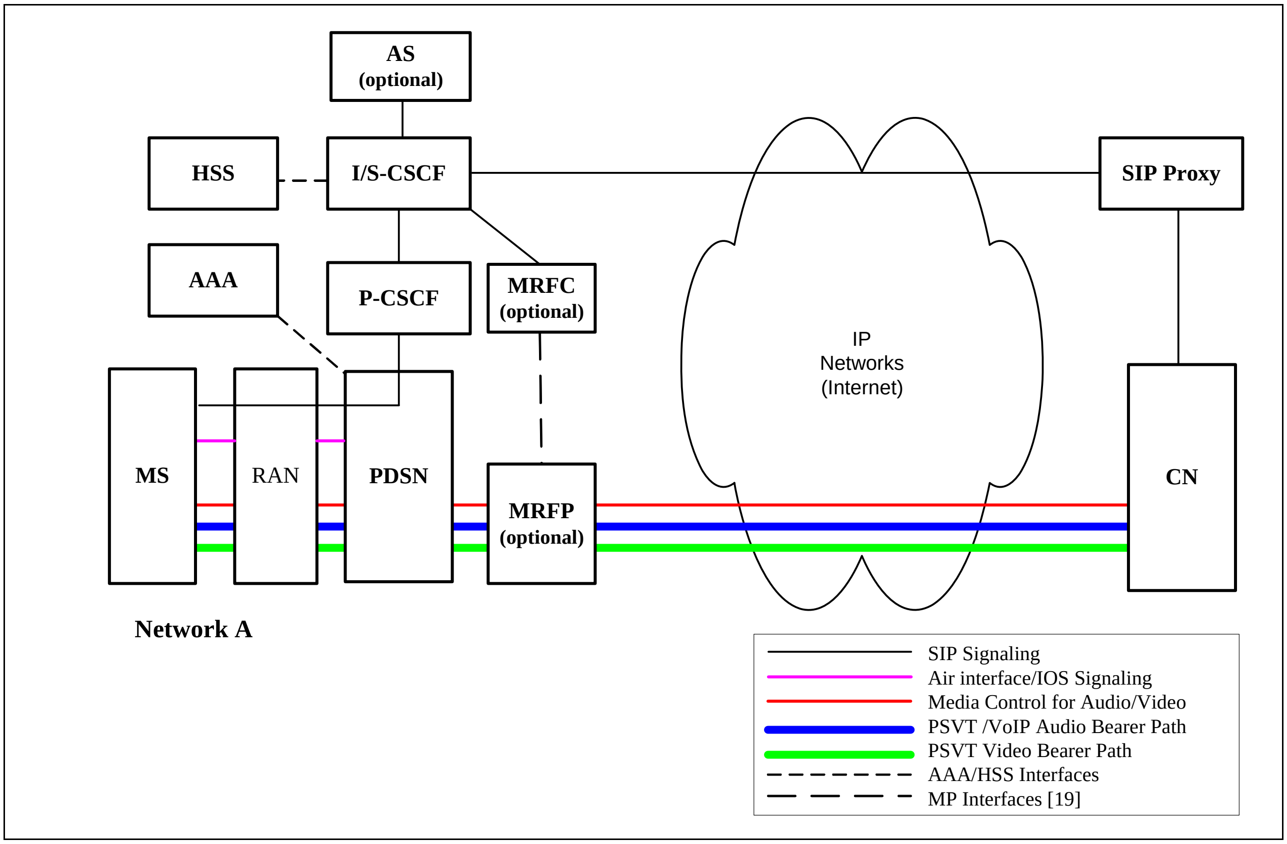

Figure 2 Network Reference Model for Non-roaming Mobile-to-Internet CN PSVT or VoIP Call

CN

The Correspondent Node is a packet switched video and/or voice terminal on the Internet. This terminal complies with the protocols and procedures above the IP level that are specified in this document.

5.2 Protocol Stack

Control and media traffic flows are carried over the IP protocol. Control traffic flow includes session/call control traffic (SIP/UDP or SIP/TCP) and media control (RTCP/UDP). Media traffic flow refers to RTP/UDP traffic. They have different protocol stacks and may travel through different paths, as illustrated in Figure 3.

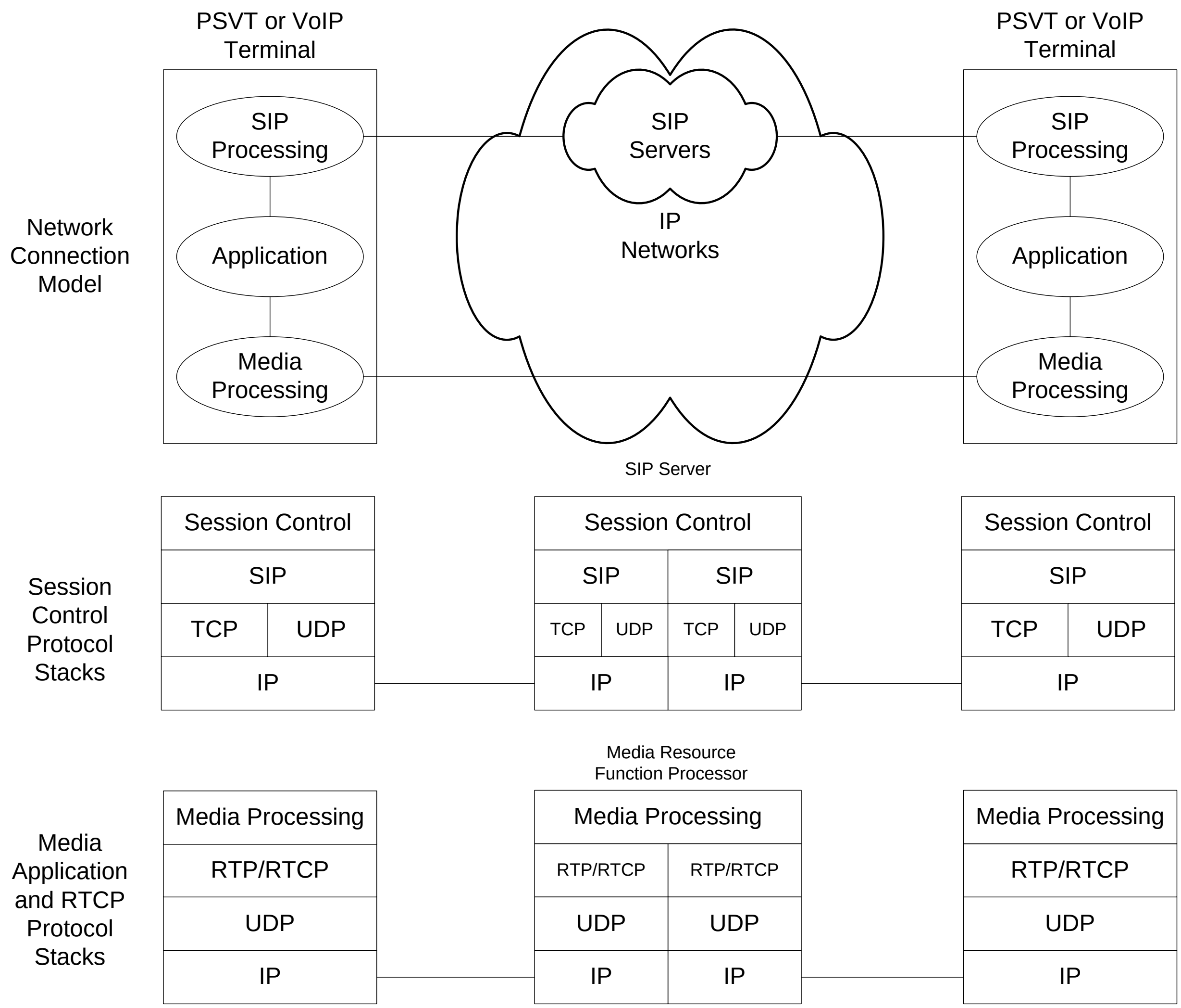

Figure 3 PSVT and VoIP Protocol Stacks over IP

Figure 4 illustrates the protocol reference model for the PSVT and VoIP Services SIP control flow between two PSVT or two VoIP mobiles when HDLC framing is used between the PDSN and MS to frame IP packets. For 1X this illustrates service option 33 or 66 over the air interface. For HRPD this illustrates service option 59 or 64.

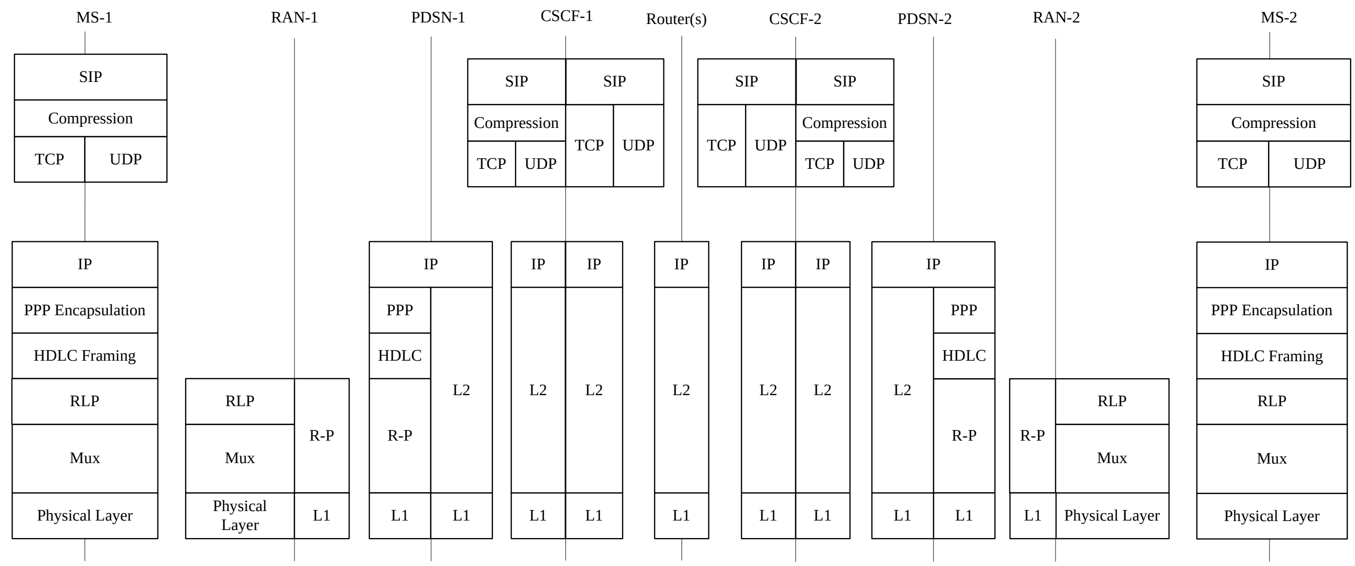

Figure 4 Protocol Stack for the SIP Control Flow (when HDLC framing is applied)

Figure 5 illustrates the protocol reference model for the PSVT or VoIP Services SIP control flow between two PSVT or two VoIP mobiles when segment-based framing (service option 67 for HRPD) is used between the RAN and MS to frame IP packets.

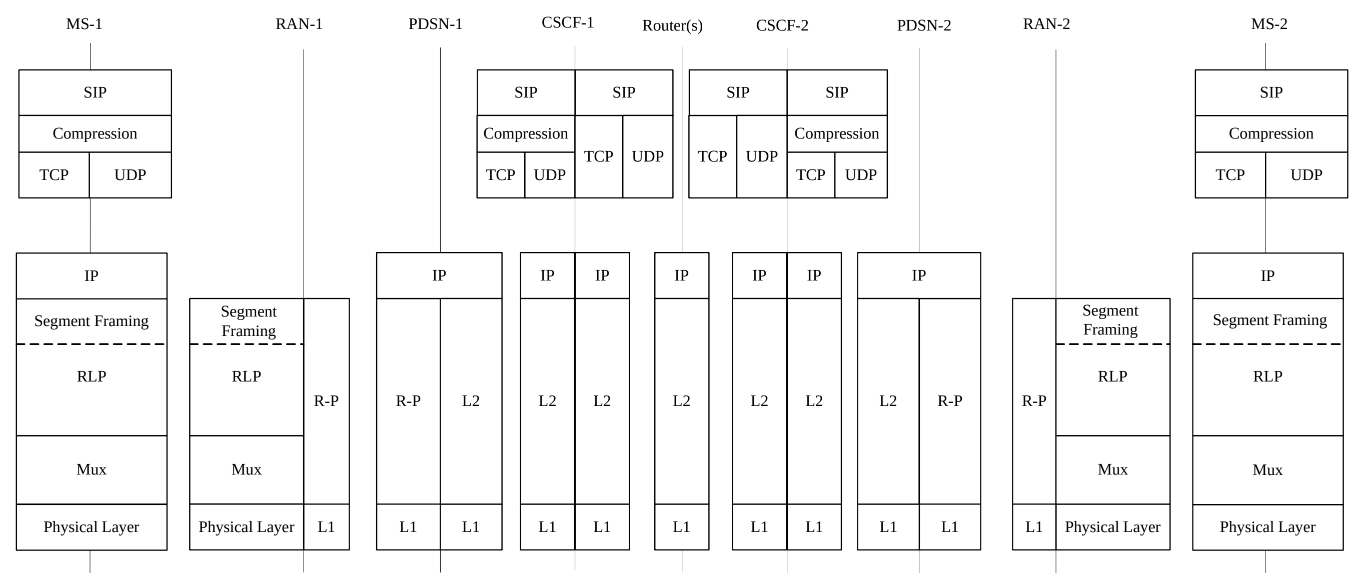

Figure 5 Protocol Stack for the SIP Control Flow (when segment-based framing is applied)

Figure 6 illustrates the protocol reference model for PSVT or VoIP Services media flows between two PSVT or two VoIP mobiles when HDLC framing is used between the PDSN and MS to frame IP packets. The protocol stack reference model for VoIP services does not include the video application component in the mobiles or MRFP. For 1X this illustrates service option 66 over the air interface. For HRPD this illustrates service option 64.

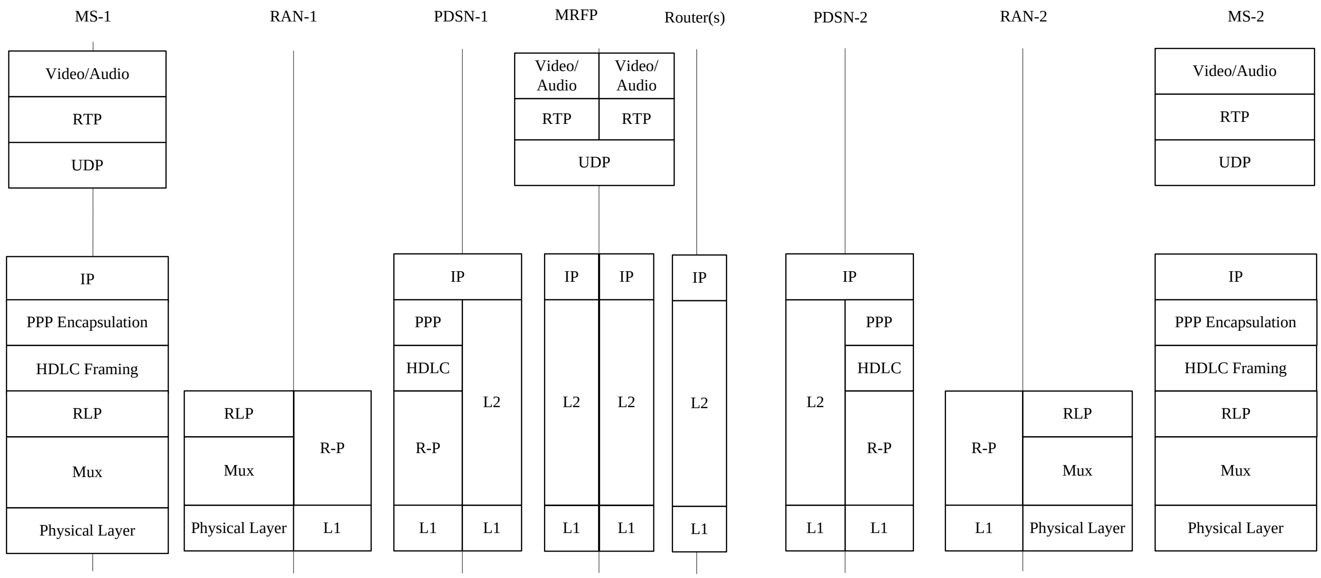

Figure 6 Protocol Stack for the Media Flows (when HDLC framing is applied)

Figure 7 illustrates the protocol reference model for the PSVT or VoIP Services media flows between two PSVT or two VoIP mobiles when segment-based framing (service option 67 for HRPD) is used between the RAN and MS to frame IP packets. The protocol stack reference model for VoIP services does not include the video application component in the mobiles or MRFP.

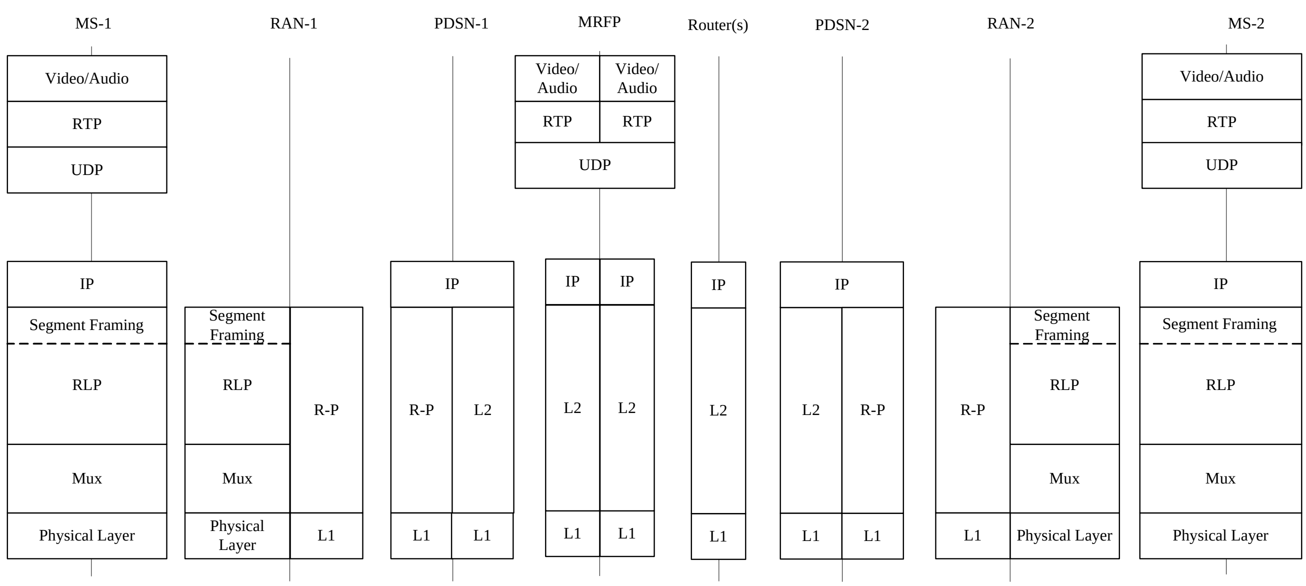

Figure 7 Protocol Stack for the Media Flows (when segment-based framing is applied)

For PSVT services, RTCP may be used by the MS to provide information needed for the synchronization of the audio and video media streams. Such synchronization is not needed for VoIP services.

6 Session Control

6.1 Session Control Signaling

6.1.1 Over-the-air Link Establishment

The MS requests establishment of an over-the-air link with the RAN to transport the SIP signalling and RTCP over the radio interface.

In 1X, the MS first requests the main packet data service option (33) which is connected to the PDSN. The MS is then assigned a Simple or Mobile IP address. Once this is granted by the RAN, SIP signaling and RTCP can be transported over this main service option. Alternatively, the MS can request auxiliary service options (66) with different QoS attributes to transport the SIP signalling and RTCP together or separately.

In HRPD, the MS first requests main link flows in the forward and reverse direction which are connected as service option 59 to the PDSN. The mobile is then assigned a Simple or Mobile IP address. SIP signalling and RTCP can be transported over these main link flows. Alternatively, the MS can request different QoS attributes for auxiliary link flows in each direction (connected as service options 64 or 67 to the PDSN) to transport the SIP signalling and RTCP together or separately.

6.1.2 A10 Connection Establishment

As the over-the-air links are connected the RAN and PDSN establish corresponding A10 connections between them. Traffic flow filters for these A10 connections are also set by the MS. If Service Based Bearer Control of MMD is not required by the network for these bearers then these connections and filters may be established prior to call setup to reduce setup delay. Otherwise, the procedures for SBBC are followed (SIP signalling bearers may be established prior to call setup even when SBBC is required by the network.)

6.1.3 IP Multimedia Session Registration

The SIP level registration can be initiated after access network registration has been completed, and the IP bearer for the signaling has been connected to the access network. The MS registers with the S-CSCF.

6.1.4 IP Multimedia Session Establishment

The PSVT and VoIP services use the IP Multimedia Subsystem [19] to establish the IP multimedia session. Figure 8 illustrates an example call flow for establishing a mobile-to-mobile IP multimedia session. The different steps in the call flow identify the main functions that need to be performed.

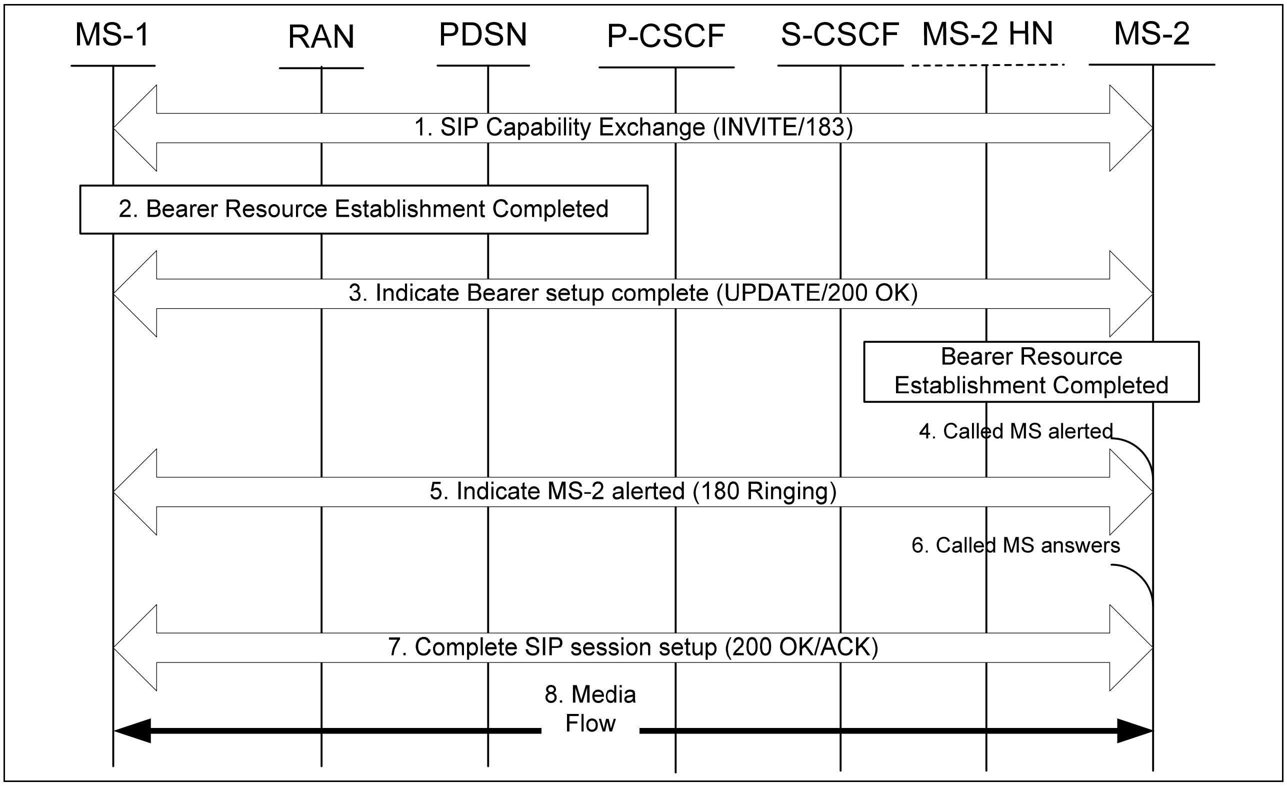

Figure 8 IP Multimedia Session Establishment - Roaming

1. When MS-1 originates a call to MS-2, MS-1 and MS-2 exchange their media capabilities through a series of SIP messages. The SDP, present in the SIP messages, contains one or more media descriptions for the multimedia session. The SIP messages traverse through the visited and the home networks of both the mobile stations.

2. Based on the session parameters agreed in the previous step, MS-1 completes establishment of bearer resources in its serving network.

   To minimize call setup time this step can be performed in advance of step 1 when Service Based Bearer Control is not required by the network. In this scenario the bearer resource reservation is not based on the agreed session parameters. Resource reservation may be based on the maximum resources that MS-1 expects to use for the IMS session.

3. After the bearer resources are set up in its serving network, MS-1 indicates to MS-2 that the reservation has been completed.

4. After MS-2 completes resource reservation in its serving network, MS-2 alerts the user.

5. MS-2 indicates to MS-1 that it is alerting the user.

6. The called party answers the phone.

7. MS-2 indicates to MS-1 that the called-party has answered and completes the call setup.

8. The PSVT or VoIP session is established and MS-1 & MS-2 now exchange media traffic. To reduce call setup time media traffic exchange can also start at step 6.

6.2 Session Media Traffic

6.2.1 Over-the-air Link Establishment

The MS requests establishment of over-the-air links with the RAN to transport media traffic over the radio interface.

In 1X, all media traffic is transported over auxiliary service options (60, 61, or 66). The MS and RAN establish the necessary service options according to one of the following procedures:

- For VoIP services the MS requests an auxiliary service option (60, 61, or 66) with the appropriate QoS for transport of the speech stream with the flow profiles described in section 7.3
- For PSVT services one of the following procedures is used:
    - The MS requests an auxiliary service option (66) with the appropriate QoS for the combined transport of both the video and audio streams with the flow profiles described in section 7.3. The RAN grants this QoS request on an existing service option or the RAN establishes a new service option instance to provide the requested QoS. Having the audio stream share a service connection with video stream is acceptable if the audio stream does not require a QoS treatment distinct from the video stream.
    - The MS establishes two auxiliary service options (66) for the separate transport of the audio and video streams with the flow profiles described in section 7.3. The RAN grants these QoS requests on existing service options or the RAN establishes new service option instances to provide the requested QoS.
      Separate transport of the media streams allows the RAN to provide different QoS treatments for the streams. Other options for the separate transport of the audio stream are service option 60 or 61. Service option 60 can be used but does not provide audio-video synchronization information (audio media RTP timestamps) to the MS. Service option 61 can be used but imposes significant implementation complexity which is not warranted by the savings of zero-byte audio packet overhead when in a PSVT call.

In HRPD, all the media traffic is transported over auxiliary link flows that are connected to the PDSN as service option 64 or 67. The MS and RAN establish the necessary auxiliary link flows according to one of the following procedures:

- For VoIP services the MS request two auxiliary link flows (one in each direction) for the transport of the speech stream with the flow profiles described in section 7.3.
- For PSVT services one of the following procedures is used:
    - The MS requests two auxiliary link flows (one in each direction) for the combined transport of both the video and audio streams with the flow profiles described in section 7.3. The RAN grants this QoS request on existing link flows or the RAN establishes new link flows to provide the requested QoS.
    - The MS establishes two pairs of auxiliary link flows for the separate transport of the audio and video streams with the flow profiles described in section 7.3. The RAN grants these QoS requests on existing link flows or the RAN establishes new link flows to provide the requested QoS. Separate transport of the media streams allows the RAN to provide different QoS treatments for the streams.

6.2.2 A10 Connection Establishment

As the over-the-air links are connected the RAN and PDSN establish corresponding A10 connections between them. Traffic flow filters for these A10 connections are also set by the MS. If Service Based Bearer Control is not required by the network then these connections and filters may be established prior to call setup to reduce setup delay. Otherwise, the Service Based Bearer Control procedures will be followed.

6.2.2.1 Terminal Addressing

The MS and network support SIP URIs and tel URLs (MDNs) as specified in [19].

6.2.2.2 PSVT Voice Retry

Due to lack of resources in the calling or called terminal's network, a MS may be unable to initiate a PSVT session. When this happens the MS attempts to connect a voice-over-IP or circuit-switched voice call to the called party if the user has selected such fall back operation.

6.3 Session Release

6.3.1 IP Multimedia Session Release

Figure 9 illustrates the call flow for a mobile initiated IP multimedia session release when one of the mobiles is roaming.

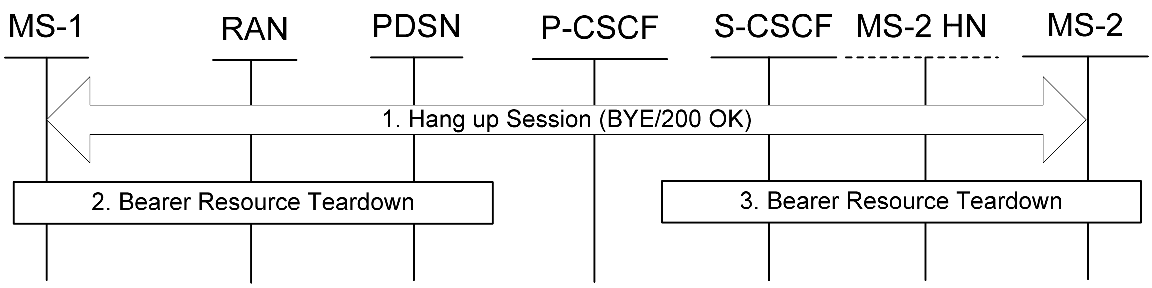

Figure 9 Mobile Initiated IP Multimedia Session Release

1. If the MS-1 wants to terminate the session, it sends a SIP BYE messages towards MS-2. The SIP messages traverse through all the proxies that are a part of the session.
2. MS-1 releases all bearer resources which were allocated to the session. This may also be performed in parallel with step 1 as long as the service connection transporting SIP control traffic is maintained.
3. MS-2 releases all bearer resources which were allocated to the session. This may also be performed in parallel with step 1 as long as the service connection transporting SIP control traffic is maintained.

6.4 In-Session Control

6.4.1 Media Stream Management

The MS supports the ability for the user to add, remove, or modify different media streams as follows:

- A new SDP offer will be used to modify the existing session attributes as described in [24].
- An MS that receives a new SDP offer notifies the user of the request to modify the session. The MS renegotiates QoS reservations as needed to support the new session attributes.
- If a media stream is on hold, the MS does not release the QoS reservation for that stream.

6.4.2 Call Waiting Notification

While the MS is in an active PSVT or VoIP session it alerts the user of other incoming calls using the following procedures:

1. If the incoming call is over IMS, the MS receives a SIP INVITE message over a 1X packet data instance or a HRPD link flow.
2. If the incoming call is from the legacy network (via the MSC or MSCe), the MS can be either
    a. A 1X-only MS which receives an alert over the 1X signaling channel, or
    b. A hybrid HRPD-1X MS which receives a page
        i. Over HRPD signaling (via cross-paging) [1] and [2], or
        ii. Over the 1X paging channel.

7 Quality of Service

7.1 Flow Treatment

The PSVT and VoIP services support the QoS requirements for each flow as specified in section 7.3. Figure 10 illustrates example QoS treatments applied by the RAN to the different data flows. For VoIP services, transport of the video RTP and video RTCP as shown in Figure 10 are not applicable.

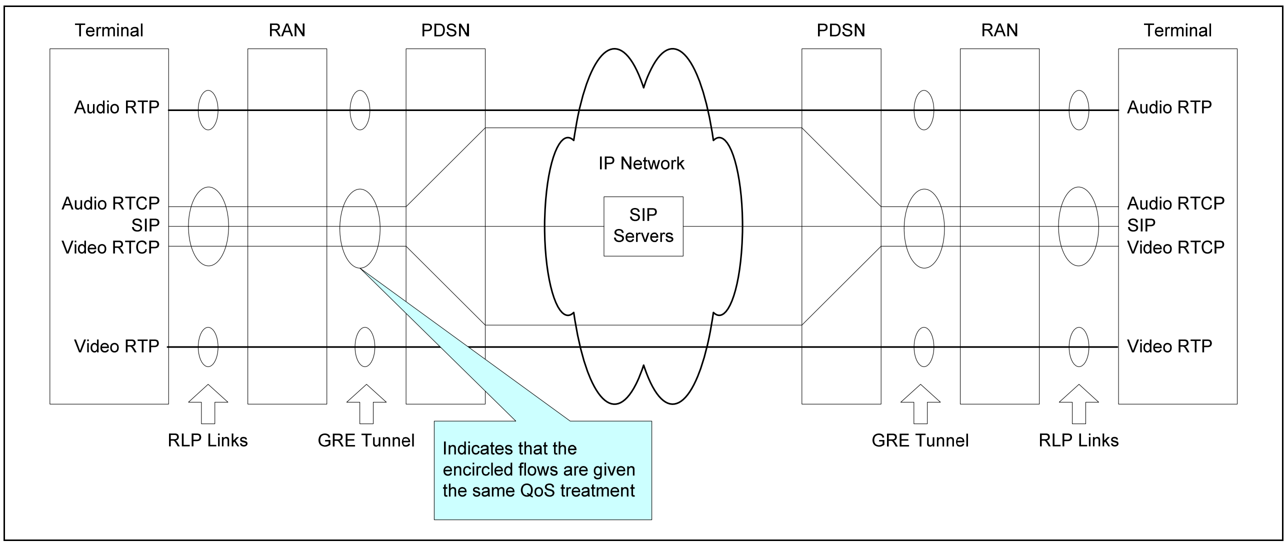

Figure 10 Access Network QoS Treatments for Different Data Flows

7.2 End-to-End QoS Transport

Figure 11 illustrates an example of how IP packets are provided QoS treatments as they are transported between the terminal, RAN, PDSN, and core network, for a mobile-to-mobile PSVT or VoIP call. The transport of traffic in the opposite direction is not shown in the figure but is the mirror image of the call flow shown.

RLP links are used to manage different QoS treatments over the air interface between the terminal and the RAN. QoS is managed between the RAN and the PDSN using A10 connections. Diffserv is an example of how to manage QoS between the PDSNs.

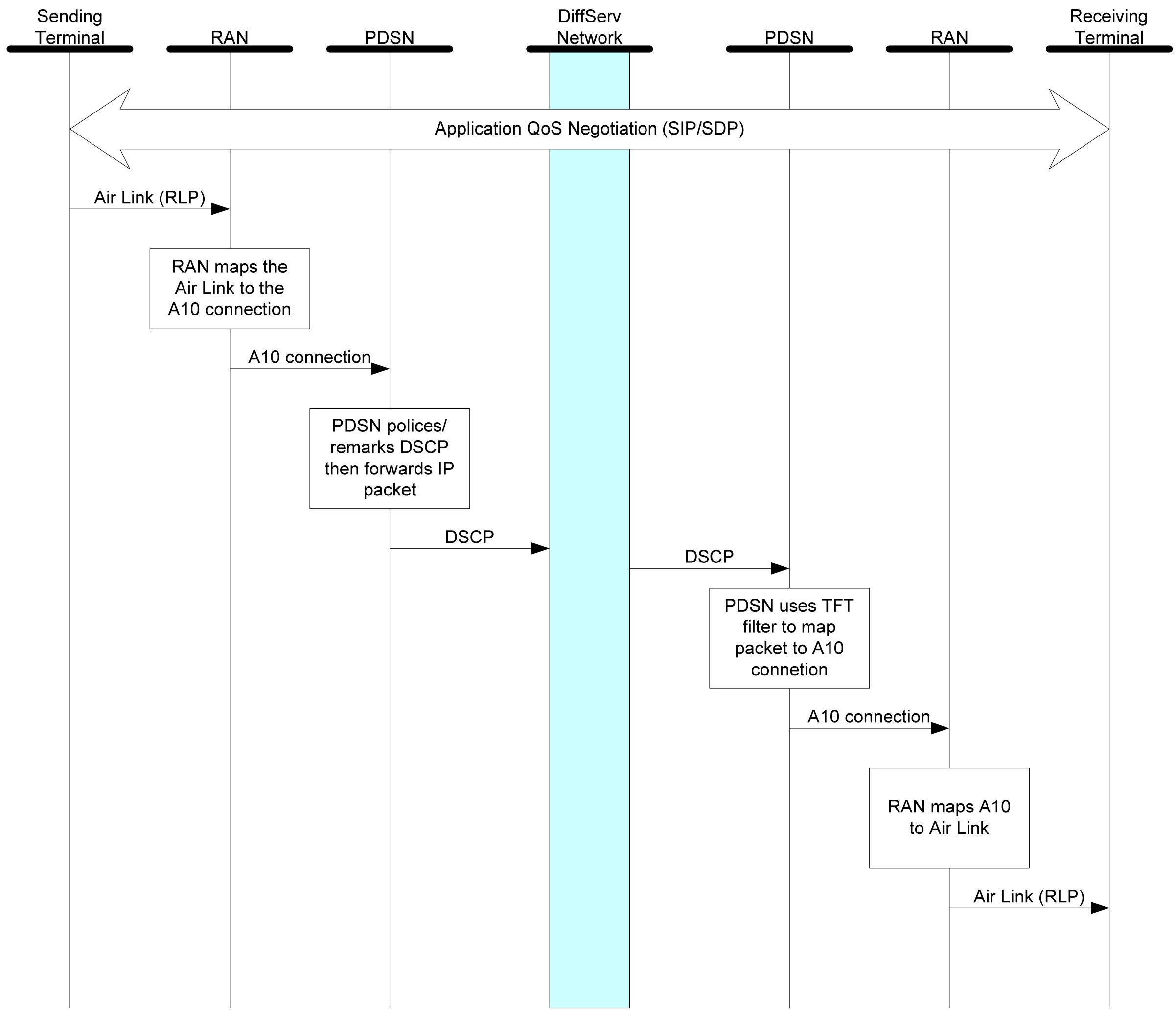

Figure 11 QoS Mappings for End-to-End PSVT/VoIP Media Transport with PDSN Policing and Remarking

For the reverse link, the MS sends session control, RTCP, and media packets on the over-the-air links with the appropriate QoS treatments. Based on the Flow Profile ID(s) and its local policy the PDSN polices and may remark the DSCP values of the reverse link packets before forwarding them to the Diffserv Network.

For the forward link, the MS sets the TFT filters at the PDSN to direct forward link traffic across the A10 flows with the appropriate QoS treatments as previously defined by the RAN. Each of the A10 flows is mapped into a corresponding over-the-air link at the RAN.

7.3 Establishing QoS

When the MS establishes the main HRPD link flow or 1X service option the PDSN obtains the Flow Profile ID(s) authorized for the user from the H-AAA and passes these to the RAN for RAN QoS authorization purposes [17].

The MS reserves a QoS treatment for each media and control flow by requesting an appropriate Flow Profile ID [5] for each flow using the procedures specified in [10] or [13]. The SIP signaling flow uses a Flow Profile ID that supports SIP signaling traffic or best effort QoS. The audio/speech flow uses a Flow Profile ID that supports audio/speech traffic. The video flow of a PSVT session uses a Flow Profile ID that supports video traffic. RTCP

traffic uses a Flow Profile ID that supports RTCP traffic or best effort QoS.
Alternatively, instead of using Flow Profile IDs, the terminal can use detailed QoS parameters to request the necessary QoS treatment for each type of stream from the RAN.

If a QoS reservation is to be used for transporting SIP signaling then the MS reserves this QoS treatment with the RAN and establish the corresponding TFT filter(s) at the PDSN before a SIP session is initiated. Before the called party is alerted of the PSVT or VoIP session the QoS reservation(s) for media flows and the setting of their corresponding TFT filters are completed by both the calling and called mobile stations with their respective RAN's and PDSNs.

7.4 QoS Session Management

The network notifies the MS of changes in the QoS reservations. The MS notifies the user of this change and provides the following options to the user

1. If the change is a reduction in QoS
    - Continue the session unmodified
    - Modify the session to adapt to the new QoS
    - Terminate the session
2. If the change is an improvement in QoS
    - Continue the session unmodified
    - Modify the session to adapt to the new QoS level

The choice of these options can be selected before the user starts a PSVT or VoIP session.

8 Compression

8.1 SIP Compression

Support for SIP compression uses the procedures specified in [20].

8.2 Header Compression

To reduce the overhead of the audio and video streams ROHC [23] can be used to compress the headers of these streams. Other compression protocols such as ECRTP [25] may also be used for service options that support the compression protocol. These compression protocols can be used in either HRPD or 1X air interfaces.

In 1X, LLA-ROHC (Service Option 61 [15]) may also be used to compress the audio stream headers over 1X channels when a native cdma2000 codec is used. Service Option 60 [15] may be used to compress the audio stream headers but will require additional procedures to provide audio-video media synchronization. These synchronization procedures are beyond the scope of this document.

The amount of overhead per audio payload frame can also be reduced by bundling multiple audio frames in a single audio RTP packet.

9 Security

9.1 Authentication and Authorization

9.1.1 Over-the-Air Link

The HLR and AN-AAA provide the authentication and authorization information for the 1X and HRPD radio links, respectively. Over-the-air access authentication and authorization of the MS is performed by the RAN as described in [10] or [13].

9.1.2 Packet Data Service/ IP Bearer

The H-AAA provides the authentication and authorization information for the packet data service/IP bearer, including authorization of Flow Profile IDs and treatments. The H-AAA performs the authentication and authorization of the MS as described in [17].

9.1.3 IP Multimedia Subsystem

The HSS provides the authentication and authorization information for the IMS. The S-CSCF performs the authentication and authorization of the MS as described in [16].

9.2 Encryption and Integrity Protection

9.2.1 Over-the-Air Link Layer Encryption

Link layer encryption over the 1X and HRPD RAN's can be provided using the procedures in [11] and [14].

9.2.2 SIP Control Signaling Security

Security of all SIP control signaling between various IMS entities uses the procedures specified in [16].

9.2.3 End-to-End Media Encryption

For further study.

10 Accounting

10.1 Offline Charging

10.1.1 Packet Data Accounting

Offline packet data accounting for PSVT or VoIP sessions is performed using the accounting procedures specified in [17].

10.1.2 MMD Offline Charging

MMD offline charging is performed using the procedures specified in [21] and [22].

10.1.3 Service Based Bearer Control Accounting

For further study.

10.2 Online Charging

10.2.1 PrePaid Packet Data Accounting

PrePaid packet data accounting for PSVT or VoIP sessions is performed using the accounting procedures for PrePaid packet data services specified in [17].

10.2.2 MMD Online Charging

For further study.

10.2.3 Service Based Bearer Control Accounting

For further study.

11 Mobility Management

11.1 Inter-BS and Inter-PDSN Handoffs

The inter-BS and inter-PDSN handoff procedures specified in [1][2][3][4][17] are used by the network to provide handoffs of the PSVT or VoIP session.

11.2 HRPD-1X Handoffs

For further study.

12 Regulatory Requirements

Support for regulatory requirements for Packet Switched Video Telephony (PSVT) and Voice-Over-IP (VoIP) services over cdma2000 are outside the scope of this technical report.