

# ETSI TS 129 199-2 V7.4.0 (2007-06)

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*Technical Specification*

**Universal Mobile Telecommunications System (UMTS);  
Open Service Access (OSA);  
Parlay X web services;  
Part 2: Third party call  
(3GPP TS 29.199-02 version 7.4.0 Release 7)**

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Reference

RTS/TSGC-0529199-02v740

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Keywords

UMTS

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## Foreword

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## Foreword

This Technical Specification has been produced by the 3<sup>rd</sup> Generation Partnership Project (3GPP).

3GPP acknowledges the contribution of the Parlay X Web Services specifications from The Parlay Group. The Parlay Group is pleased to see 3GPP acknowledge and publish the present document, and the Parlay Group looks forward to working with the 3GPP community to improve future versions of the present document.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

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## Introduction

The present document is part 2 of a multi-part deliverable covering the 3<sup>rd</sup> Generation Partnership Project; Technical Specification Group Core Network and Terminals; Open Service Access (OSA); Parlay X Web Services, as identified below:

Part 1:	"Common"
<b>Part 2:</b>	<b>"Third party call"</b>
Part 3:	"Call Notification"
Part 4:	"Short Messaging"
Part 5:	"Multimedia Messaging"
Part 6:	"Payment"
Part 7:	"Account management"
Part 8:	"Terminal Status"
Part 9:	"Terminal location"
Part 10:	"Call handling"
Part 11:	"Audio call"
Part 12:	"Multimedia conference"
Part 13:	"Address list management"
Part 14:	"Presence"
Part 15:	"Message Broadcast"
Part 16:	"Geocoding"
Part 17:	"Application driven Quality of Service (QoS)"
Part 18:	"Device Capabilities and Configuration"
Part 19:	"Multimedia streaming control"
Part 20:	"Multimedia multicast session management"

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# 1 Scope

The present document is Part 2 of the Stage 3 Parlay X Web Services specification for Open Service Access (OSA).

The OSA specifications define an architecture that enables application developers to make use of network functionality through an open standardized interface, i.e. the OSA APIs. The concepts and the functional architecture for the OSA are contained in 3GPP TS 23.127 [3]. The requirements for OSA are contained in 3GPP TS 22.127 [2].

The present document specifies the Third Party Call Web Service aspects of the interface. All aspects of the Third Party Call Web Service are defined here, these being:

- Name spaces.
- Sequence diagrams.
- Data definitions.
- Interface specification plus detailed method descriptions.
- Fault definitions.
- Service Policies.
- WSDL Description of the interfaces.

The present document has been defined jointly between 3GPP TSG CT WG5, ETSI TISPAN and The Parlay Group.

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# 2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present 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 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

- [1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".
- [2] 3GPP TS 22.127: "Service Requirement for the Open Services Access (OSA); Stage 1".
- [3] 3GPP TS 23.127: "Virtual Home Environment (VHE) / Open Service Access (OSA); Stage 2".
- [4] 3GPP TS 22.101: "Service aspects; Service principles".
- [5] W3C Recommendation (2 May 2001): "XML Schema Part 2: Datatypes".

NOTE: Available at <http://www.w3.org/TR/2001/REC-xmlschema-2-20010502/>.

- [6] 3GPP TS 29.199-1: "Open Service Access (OSA); Parlay X Web Services; Part 1: Common".
- [7] 3GPP TS 29.199-12: "Open Service Access (OSA); Parlay X Web Services; Part 12: Multimedia conference'.

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# 3 Definitions and abbreviations

## 3.1 Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TS 29.199-1 [6] apply.

## 3.2 Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP TS 29.199-1 [6] apply.

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# 4 Detailed service description

Currently, in order to perform a third party call in telecommunication networks we have to write applications using specific protocols to access Call Control functions provided by network elements (specifically operations to initiate a call from applications). This approach requires a high degree of network expertise. We can also use the OSA gateway approach, invoking standard interfaces to gain access to call control capabilities, but these interfaces are usually perceived to be quite complex by application IT developers. Developers must have advanced telecommunication skills to use Call Control OSA interfaces.

In this subclause we describe a Parlay X Web Service, Third Party Call, for creating and managing a call initiated by an application (third party call). The overall scope of this Web Service is to provide functions to application developers to create a call in a simple way. Using the Third Party Call Web Service, application developers can invoke call handling functions without detailed telecommunication knowledge.

The underlying model of the service is based on the following entities:

- **Call Session:** a call (uniquely identified) to which participants can be added/removed.
- **Call Participant:** each of the call parties (uniquely identified) involved in the call session.
- **Media:** the call can utilize multiple media types to support the participants' communication. In particular both audio and video streams are available, including the specific stream direction (i.e. incoming, outgoing, bidirectional).

NOTE 1: Call participants in a Call Session are anticipated to be uniquely identifiable using their URI address.

An application setting up a call session must initially invoke the **makeCallSession**. The result of such invocation is the creation of a "context" that represents a call session with usually two participants, or at a minimum one participant connected, a unique identifier is assigned to the just-created call session. Subsequently the application may wish to add, remove, park or transfer call participants. In order to do so the operations **addCallParticipant**, **transferCallParticipant**, **deleteCallParticipant** can be used. Furthermore the call session or call participant status including the media details can be read. In order to do so the operations **getCallParticipantInformation**, and **getCallSessionInformation** can be used. It is also possible to retrieve the media details on its own using the **getMediaForParticipant** or **getMediaForCall** operations of the Audio Call web service.

The application can also force the call session and all its participants to be terminated with the operation **endCallSession**.

NOTE 2: A call session allows the application to avail of other web service features that can add value to the created call session. For example the Audio Call web service can provide multimedia message delivery to call participants in the call session (playXXXMessage operation) and furthermore control of the media types for the call participants thus enabling conversational multimedia communication including voice, video, chat, and data. Media can be added/removed for each participant.

Figure 4.1 shows a scenario using the Third Party Call Web Service to handle third party call functions. The application invokes a Web Service to retrieve stock quotes and a Parlay X Interface to initiate a third party call between a broker and his client.

In the scenario, whenever a particular stock quote reaches a threshold value (1) and (2), the client application invokes a third party call between one or more brokers and their corresponding customers to decide actions to be taken. After invocation (3) by the application, the Third Party Call Web Service invokes a Parlay API operation (4) using the Parlay/OSA SCS-CC (Call control) interface. This SCS handles the invocation and sends a message (5) to an MSC to set-up a call between user A and user B.

In an alternative scenario, the Parlay API interaction involving steps (4) and (5) could be replaced with a direct interaction between the Third Party Call Web Service and the Mobile network



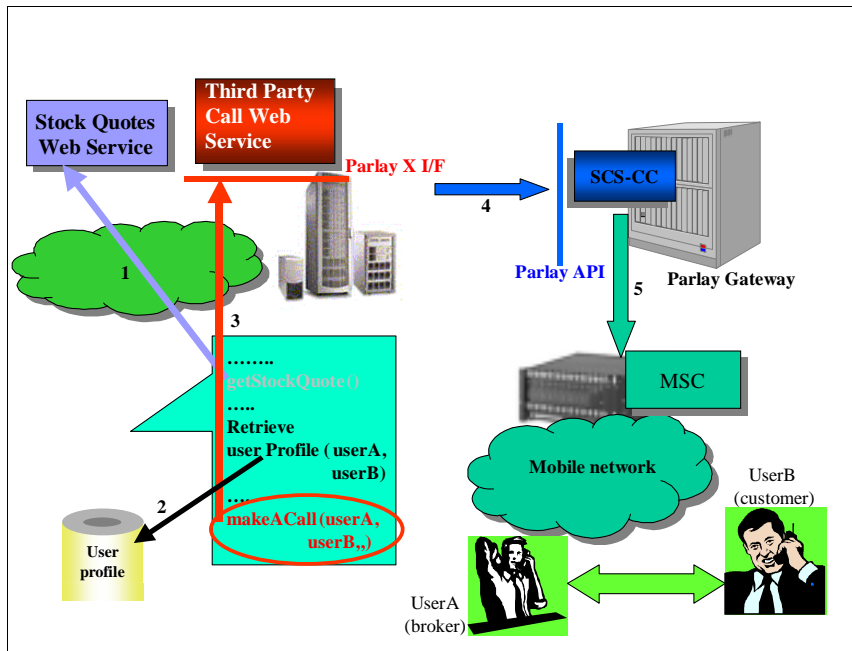


Figure 4.1: Third party call scenario

## 5 Namespaces

The ThirdPartyCall interface uses the namespace:

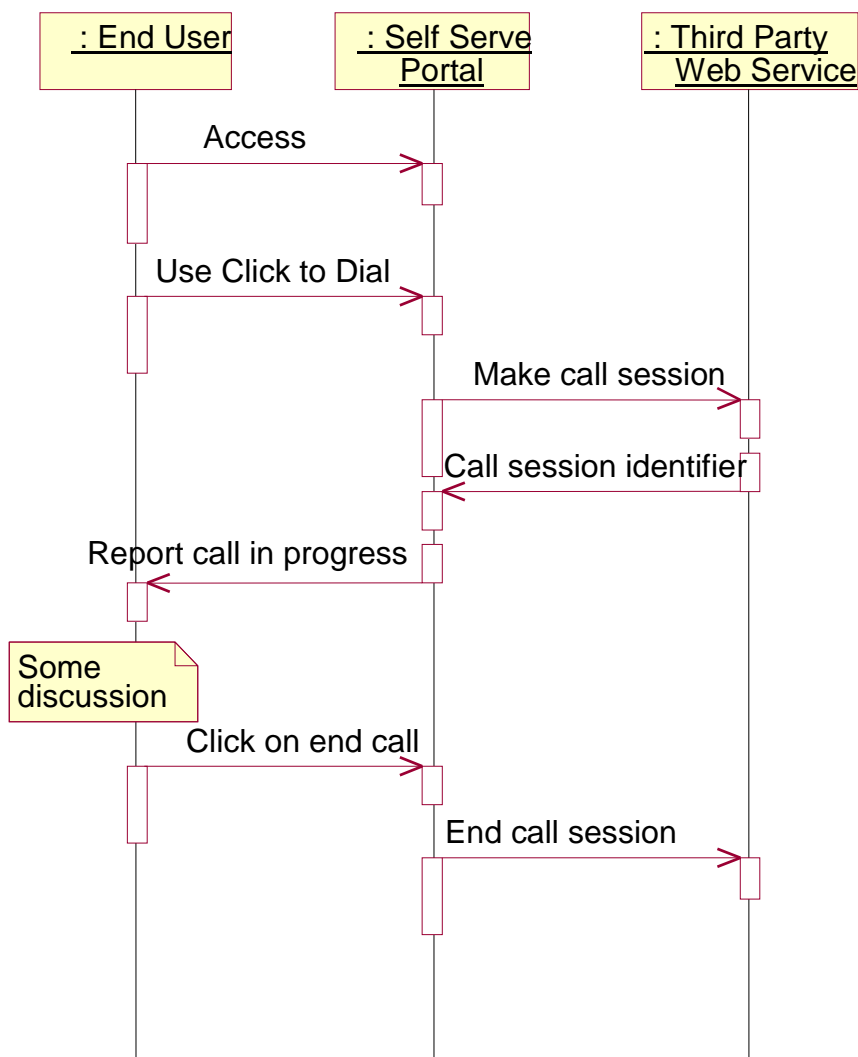
[http://www.csapi.org/wsd/parlayx/third\\_party\\_call/v3\\_4](http://www.csapi.org/wsd/parlayx/third_party_call/v3_4)

The 'xsd' namespace is used in the present document to refer to the XML Schema data types defined in XML Schema [5]. The use of the name 'xsd' is not semantically significant.

## 6 Sequence diagrams

### 6.1 'Click to Dial' call setup

A common convergence application is Click to Dial, where a self service portal provides a web page that can initiate a call between two phones. This sequence shows a basic call setup, and ending the call through the portal.



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## 7 XML Schema data type definition

Void.

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## 8 Web Service interface definition

### 8.1 Interface: ThirdPartyCall

This interface provides the ability to setup a call session, add and delete a call participant, transfer a call participant from one call session into another call session, determine the status of an individual call participant or a complete call session, and finally to end a call session.

#### 8.1.1 Operation: MakeCallSession

The invocation of **MakeCallSession** requests to set-up a call session between two addresses, a first **CallParticipant** ('A-Party') and a second **CallParticipant** ('B-Party'), provided that the invoking application is allowed to connect them. Optionally, the caller's ('A-Party's') name is provided. Optionally a call session with only a single **CallParticipant** ('A-party') can be set-up, for example to play media to the call participant using mediaCall capabilities (e.g. playMedia). Optionally the application can also indicate the charging information (**Charging**). Optionally, the media type(s) to be used for the participants in the call session can be requested (**MediaInfo**). A service policy details if multimedia application control is supported. If the parameter **MediaInfo** is omitted, the media type(s) shall be negotiated by the underlying network. A **CallSessionIdentifier** is returned on invocation of the **MakeCallSession**. This **CallSessionIdentifier** may be used to retrieve the call session status for all the call participants including their media type information with the operation **getCallSessionInformation**. With the operation **getCallParticipantInformation** this information may be retrieved for a single call participant.

During call set-up, the first **CallParticipant** ('A-Party') might wait for the second **CallParticipant** ('B-Party') to answer the call hearing ringback tone. **MakeCallSession** creates a call session with one or two call participants and generates a new **CallSessionIdentifier** which is identified by the string returned in the **MakeCallSessionResponse** operation. The invocation returns as soon as the request is received by the system, i.e. the actual call session is established asynchronously.

In order to receive the information on call status the application has to explicitly invoke the **GetCallSessionInformation** operation for the **CallSessionIdentifier** returned.

### 8.1.1.1 Input message: MakeCallSessionRequest

Part name	Part type	Optional	Description
CallParticipants	xsd:anyURI[1..2]	No	It contains the address of the first participant, and if supplied the second participant, involved in the call session.
CallingParticipantName	xsd:string	Yes	It contains the name of the caller, e.g. the name on whose behalf the call session is being established.
Charging	common:ChargingInformation	Yes	Charge to apply to the call session
MediaInfo	common:MediaInfo [0..unbounded]	Yes	It identifies one or more media type(s) for the call, i.e. the media type(s) to be applied to the participants in the call session. For each media type the media direction: incoming, outgoing, or bidirectional shall be indicated. An empty array shall have the same meaning as if the parameter is omitted. If the parameter <b>MediaInfo</b> is omitted, the media type(s) shall be negotiated by the underlying network.
ChangeMediaNotAllowed	xsd:boolean	No	If true, no call participant (user) in the call will be permitted to change media type during the call. If false the end user may change media type after the call is established as no network protection mechanism is set up to prevent participant (end user) initiated change of media type.

### 8.1.1.2 Output message : MakeCallSessionResponse

Part name	Part type	Optional	Description
CallSessionIdentifier	xsd:string	No	It identifies the specific call session created

### 8.1.1.3 Referenced faults

ServiceException from 3GPP TS 29.199-1 [6]:

- SVC0001 - Service error.
- SVC0002 - Invalid input value.

PolicyException from 3GPP TS 29.199-1 [6]:

- POL0001 - Policy error.
- POL0008 - Charging not supported.
- POL0011 – Media type not supported

### 8.1.1a Operation: AddCallParticipant

The invocation of **AddCallParticipant** requests to add a call participant to an existing call session.

The **AddCallParticipant** operation takes two mandatory parameters, the first being the call session identifier to which the participants should be added, and the second being the participant to add (contains the URI of the participant).

Optionally, the media type(s) that shall be added for the specific call participant may be requested using **MediaInfo**.

If the parameter **MediaInfo** is omitted, the media type(s) shall be negotiated by the underlying network. The call session or call participant status, including information on media types, can be retrieved using the operations **getCallParticipantInformation**, and **getCallSessionInformation**.

All occurrences of invalid call session or participant address shall result in an invalid input value exception.

### 8.1.1a.1 Input Message : AddCallParticipantRequest

Part name	Part type	Optional	Description
CallSessionIdentifier	xsd:string	No	It identifies the existing call session. This must be a non-null value as it identifies a pre-existing call session in the network
CallParticipant	xsd:anyURI	No	It contains the address of the user to add to the existing call session identified by the CallSessionIdentifier
MediaInfo	common:MediaInfo [0..unbounded]	Yes	It identifies one or more media type(s) for the participant to be added in the call session. For each media type the media direction: incoming, outgoing, or bidirectional shall be indicated. An empty array shall have the same meaning as if the parameter is omitted. If the parameter <b>MediaInfo</b> is omitted, the media type(s) shall be negotiated by the underlying network.

### 8.1.1a.2 Output message : AddCallParticipantResponse

Part name	Part type	Optional	Description
None			

### 8.1.1a.3 Referenced faults

ServiceException from 3GPP TS 29.199-1 [6]:

- SVC0001 - Service error.
- SVC0002 - Invalid input value.
- SVC0261 - Call Session already terminated.

PolicyException from 3GPP TS 29.199-1 [6]:

- POL0001 - Policy error.
- POL0011 – Media type not supported
- POL0240 – Too many participants. - from 3GPP TS 29.199-12 [7]

### 8.1.1b Operation: TransferCallParticipant

The invocation of **TransferCallParticipant** enables a call transfer, effectively transferring a call participant from one call session into another call session.

The **TransferCallParticipant** method takes three parameters, the first being the call session identifier (destination call session) where the participants should be moved to and the second being the source call session where the participant to transfer is currently located. The third parameter is to identify the call participant to transfer (contains the URL of the participant). Upon completion of the **TransferCallParticipant** operation, the call participant is implicitly deleted from the source call session.

In transferring the call participant to the new destination call session, the participant information is not affected. When, as a result of transferring a call participant, the source call session is left without any call participant, the source call session is terminated.

All occurrences of invalid call session or participant address shall result in an invalid input value exception

### 8.1.1b.1 Input Message : TransferCallParticipantRequest

Part name	Part type	Optional	Description
DestinationCallSessionIdentifier	xsd:string	No	It identifies the existing target call session to which the call participant to transfer shall be moved.
SourceCallSessionIdentifier	xsd:string	No	It identifies the existing source call session, which contains the call participant to transfer.
CallParticipant	xsd:anyURI	No	This is the address of the participant to be transferred from the source call session, which is identified by the SourceCallSessionIdentifier.

### 8.1.1b.2 Output message : TransferCallParticipantResponse

Part name	Part type	Optional	Description
None			

### 8.1.1b.3 Referenced faults

ServiceException from 3GPP TS 29.199-1 [6]:

- SVC0001 - Service error.
- SVC0002 - Invalid input value.
- SVC0261 - Call Session already terminated.

PolicyException from 3GPP TS 29.199-1 [6]:

- POL0001 - Policy error.
- POL0240 – Too many participants. - from 3GPP TS 29.199-12 [7]

## 8.1.2 Operation: GetCallParticipantInformation

The invocation of **GetCallParticipantInformation** retrieves the current call participant status, **CallParticipantInformation**, of the call participant identified by **CallParticipant**, within the call session identified by the **CallSessionIdentifier**. This method can be invoked multiple times by the application even if the call has already ended. However, after the call has ended, status information will be available only for a limited period of time that is specified in the service policy 'StatusRetentionTime'.

The **getCallParticipantInformation** method takes two parameters into the method, the first being the call session identifier where the participants information should be retrieved and the second being the call participant identifier (contains the URL of the participant).

All occurrences of invalid call session or call participant address shall result in an invalid input value exception

### 8.1.2.1 Input message: GetCallParticipantInformationRequest

Part name	Part type	Optional	Description
CallSessionIdentifier	xsd: string	No	It identifies the specific call session
CallParticipant	xsd:anyURI	No	It identifies a specific call participant address within the call session

### 8.1.2.2 Output message : GetCallParticipantInformationResponse

Part name	Part type	Optional	Description
result	common: CallParticipantInformation	No	It identifies the status of the requested call participant

### 8.1.2.3 Referenced faults

ServiceException from 3GPP TS 29.199-1 [6]:

- SVC0001 - Service error.
- SVC0002 - Invalid input value.
- SVC0261 - Call Session already terminated.

PolicyException from 3GPP TS 29.199-1 [6]:

- POL0001 - Policy error.

### 8.1.3 Void

### 8.1.4 Void

### 8.1.5 Operation: GetCallSessionInformation

The invocation of **GetCallSessionInformation** retrieves the **CallSession** information regarding the call session identified by **CallSessionIdentifier** parameter. The information provided in the **CallSession** includes the call participant information. This method can be invoked multiple times by the application even if the session has already ended. However, after the session has ended, status information will be available only for a limited period of time that is specified in the service policy 'StatusRetentionTime'.

All occurrences of invalid call session shall result in an invalid input value exception

#### 8.1.5.1 Input message: GetCallSessionInformationRequest

Part name	Part type	Optional	Description
CallSessionIdentifier	xsd: string	No	It identifies a specific call session

#### 8.1.5.2 Output message : GetCallSessionInformationResponse

Part name	Part type	Optional	Description
result	common: CallSession	No	It identifies the status of the session

### 8.1.5.3 Referenced faults

ServiceException from 3GPP TS 29.199-1 [6]:

- SVC0001 - Service error.
- SVC0002 - Invalid input value.
- SVC0261 - Call Session already terminated.

PolicyException from 3GPP TS 29.199-1 [6]:

- POL0001 - Policy error.

### 8.1.6 Operation: DeleteCallParticipant

The invocation of **DeleteCallParticipant** removes the call participant identified by **CallParticipant** from the call session identified by **CallSessionIdentifier**, and implicitly terminates that participants involvement in the call session.

The **deleteCallParticipant** method takes two parameters, the first being the call session identifier where the participant to delete is located. The second parameter identifies the call participant to be deleted (contains the URL of the participant).

All occurrences of invalid call session or participant address shall result in an invalid input value exception

#### 8.1.6.1 Input message: DeleteCallParticipantRequest

Part name	Part type	Optional	Description
CallSessionIdentifier	xsd: string	No	It identifies a specific call session.
CallParticipant	xsd: anyURI	No	It identifies a specific call participant within the identified call session.

#### 8.1.6.2 Output message: DeleteCallParticipantResponse

Part name	Part type	Optional	Description
None			

#### 8.1.6.3 Referenced faults

ServiceException from 3GPP TS 29.199-1 [6]:

- SVC0001 - Service error.
- SVC0002 - Invalid input value.
- SVC0261 - Call Session already terminated.

PolicyException from 3GPP TS 29.199-1 [6]:

- POL0001 - Policy error.

### 8.1.7 Operation: EndCallSession

The invocation of **EndCallSession** terminates the call session identified by **CallSessionIdentifier**.

The call to all participants is ended.

All occurrences of invalid call session shall result in an invalid input value exception

#### 8.1.7.1 Input message: EndCallSessionRequest

Part name	Part type	Optional	Description
CallSessionIdentifier	xsd: string	No	It identifies a specific call session

#### 8.1.7.2 Output message: EndCallSessionResponse

Part name	Part type	Optional	Description
None			

#### 8.1.7.3 Referenced faults

ServiceException from 3GPP TS 29.199-1 [6]:

- SVC0001 - Service error.
- SVC0002 - Invalid input value.



- SVC0261 - Call session already terminated.

PolicyException from 3GPP TS 29.199-1 [6]:

- POL0001 - Policy error.

## 9 Fault definitions

The following faults are defined for this service.

### 9.1 ServiceException

#### 9.1.1 SVC0260: Void

#### 9.1.2 SVC0261: Call Session already terminated

Part name	Description
Message Id	SVC0261
Text	Call session has already been terminated
Variables	None

## 10 Service policies

These service policies are defined for the Third Party Call service.

Name	Type	Description
ChargingAllowed	xsd:boolean	Indicates whether charging is allowed for makeCallSession operation
StatusRetentionTime	common:TimeMetric	Length of time to retain status after the termination of the call
ChangeMediaNotAllowed	xsd:boolean	Indicates whether an end user can change the media types used in a call.
MaximumParticipants	xsd:int	Maximum number of participants which a session can serve.
MultimediaSupported	xsd:boolean	Indicates whether multimedia is supported and whether an application can change the media types used in a call.

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## Annex A (normative): WSDL for Third Party Call

The document/literal WSDL representation of this interface specification is compliant to 3GPP TS 29.199-1 [6] and is contained in text files (contained in archive 29199-02-740-doclit.zip) which accompanies the present document.

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## Annex B (informative): Description of Parlay X Web Services Part 2: Third party call for 3GPP2 cdma2000 networks

This annex is intended to define the OSA Parlay X Web Services Stage 3 interface definitions and it provides the complete OSA specifications. It is an extension of OSA Parlay X Web Services specifications capabilities to enable operation in cdma2000 systems environment. They are in alignment with 3GPP2 Stage 1 requirements and Stage 2 architecture defined in:

- [1] 3GPP2 X.S0011-D: 'cdma2000 Wireless IP Network Standard ', Version 1.1
- [2] 3GPP2 S.R0037-0: "IP Network Architecture Model for cdma2000 Spread Spectrum Systems", Version 3.0
- [3] 3GPP2 X.S0013-A: "All-IP Core Network Multimedia Domain"

These requirements are expressed as additions to and/or exclusions from the 3GPP Release 7 specification. The information given here is to be used by developers in 3GPP2 cdma2000 network architecture to interpret the 3GPP OSA specifications.

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### B.1 General Exceptions

The terms 3GPP and UMTS are not applicable for the cdma2000 family of standards. Nevertheless these terms are used (3GPP TR 21.905) mostly in the broader sense of "3G Wireless System". If not stated otherwise there are no additions or exclusions required.

CAMEL mappings are not applicable for cdma2000 systems.

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### B.2 Specific Exceptions

#### B.2.1 Clause 1: Scope

There are no additions or exclusions.

#### B.2.2 Clause 2: References

There are no additions or exclusions.

#### B.2.3 Clause 3: Definitions and abbreviations

There are no additions or exclusions.

#### B.2.4 Clause 4: Detailed service description

There are no additions or exclusions.

#### B.2.5 Clause 5: Namespaces

There are no additions or exclusions.

## B.2.6 Clause 6: Sequence diagrams

There are no additions or exclusions.

## B.2.7 Clause 7: XML Schema data type definition

There are no additions or exclusions.

## B.2.8 Clause 8: Web Service interface definition

There are no additions or exclusions.

## B.2.9 Clause 9: Fault definitions

There are no additions or exclusions.

## B.2.10 Clause 10: Service policies

There are no additions or exclusions.

## B.2.11 Annex A (normative): WSDL for Third Party Call

There are no additions or exclusions.

## Annex C (informative): Change history

Change history								
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Cat	Old	New
Dec 2005	CT_30	CP-050582	0003	--	Modify TPC to allow playing announcement	C	6.2.0	7.0.0
Jun 2006	CT_32	CP-060208	0006	--	Apply Union data type element naming convention	F	7.0.0	7.1.0
Dec 2006	CT_34	CP-060598	0007	--	Add Extended Call Control for Parlay X	C	7.1.0	7.2.0
Mar 2007	CT_35	CP-070045	0009	--	Add OSA Parlay Web Services support for 3GPP2 networks	A	7.2.0	7.3.0
Mar 2007	CT_35	CP-070048	0010	--	Add support for multimedia in Third party call	B	7.2.0	7.3.0
Mar 2007	--	--	--	--	Editorial: Aligned 5 Namespaces	--	7.3.0	7.3.1
Jun 2007	CT_36	CP-070346	0012	--	Correction to Third Party Call Web Service	F	7.3.1	7.4.0
Jun 2007	CT_36	CP-070346	0013	--	Refine Third Party Call Web Service	F	7.3.1	7.4.0

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## History

<b>Document history</b>		
V7.3.0	March 2007	Publication (Withdrawn)
V7.3.1	March 2007	Publication
V7.4.0	June 2007	Publication