Dialogic® PowerMedia™ XMS RESTful API

User's Guide

April 2016 05-2703-006

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Revision History

Revision	Release Date	Notes
05-2703-006 (Updated)	April 2016	Removed WebRTC support.
05-2703-006 (Updated)	December 2014	Updates to support PowerMedia XMS Release 2.2 Service Update 5. Event Handler Resource: • Added RESTful Event Streaming Data Format Change section. Events: • Updated end_play, end_playcollect, end_record, and end_playrecord sections in Media Events. • Updated end_recognize section in MRCP Events. • Updated sections to change call_id to id in MRCP Events.
05-2703-006 (Updated)	October 2014	Updates to include extra escape code for SDP in Single Call Resource section. Updates to include details for default in API Resources section. Updates to end_play event in Media Events section. Updates to connected event in Call Events section.
05-2703-006	May 2014	Updates to dtmf event in Call Events section.
05-2703-005	April 2014	Updates to support PowerMedia XMS Release 2.2.
05-2703-004	October 2013	Updates to support PowerMedia XMS Release 2.1. Global change: • Renamed this document from Developer's Guide to User's Guide. • Reorganized the resource-based elements and payload attributes tables. XML Schema Definition of Elements: • Added updated schema definition.

<table>
<tr><th>Revision</th><th>Release Date</th><th>Notes</th></tr>
<tr><td>05-2703-003</td><td>January 2013</td><td>Updates to support PowerMedia XMS Release 2.0.
Call Resource:
Updated http post/put request payload, single call instance response payload, call_action, add_party, and update_party in XML Schema Definitions for Call.
Added new clamp_dtmf, auto_gain_control, echo_cancellation, digits, interval, level, content_type, and content to the <call_action> Parameters table.
Added new send_dtmf, send_info, send_info_ack, transfer, redirect, and hangup to the <call_action> Parameters table.
Added new display_name, accept, early_media, and info_ack_mode to the <call> Element Attributes table.
Added new section for <call> Element Attribute Notes.
Conference Resource:
Added new playrecord conference action in XML Schema Definition for Conference.
Added new barge, cleardigits, beep, and recording_uri to the <conf_action> Parameters table.
XML Schema Definition of Elements:
Added updated schema definition.</td></tr>
</table>

Revision	Release Date	Notes
05-2703-002	July 2012	Updated to support PowerMedia XMS Release 1.1. This includes adding additional information to all sections and reorganizing the layout of the Resource sections. Global change: • Renamed PowerMedia XMS RESTful web service to PowerMedia XMS RESTful server. Call Resource: • Added new interdigit_timeout parameter to the <call_action> Parameters table. Conference Resource: • Added new region parameter to the <conf_action> Parameters table. XML Schema Definition of Elements: • Added updated schema definition. Dynamic Text and Image Generation: • Added new section. XMSTool RESTful Utility: • Added new section.
05-2703-001	February 2012	Initial release of this document.
Last modified: April 2016		

Refer to www.dialogic.com for product updates and for information about support policies, warranty information, and service offerings.

1. Welcome

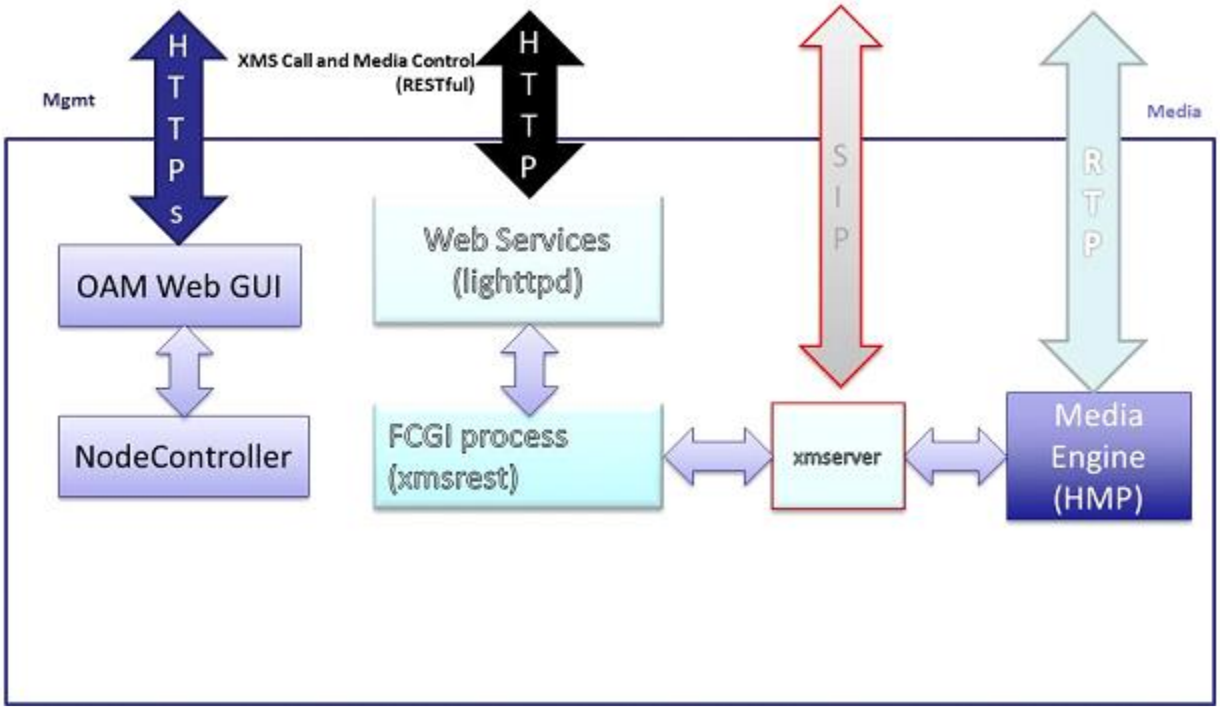
This User's Guide provides information about the Dialogic® PowerMedia™ Extended Media Server (also referred to herein as "PowerMedia XMS" or "XMS") RESTful API interface, including available features and resource-based component definitions.

The PowerMedia XMS RESTful API is one of several APIs that can be used to drive the PowerMedia XMS.

2. Overview

This section provides information about the PowerMedia XMS RESTful API interface, including available features and resource-based component definitions.

The PowerMedia XMS RESTful API is one of several APIs that can be used to drive the PowerMedia XMS. The architectural diagram below shows how the RESTful interface fits into PowerMedia XMS.



Two web servers are used in PowerMedia XMS:

- Apache (httpd) server
  Controls a web-based interface for operations, administration and maintenance.
- lighttpd server
  Processes call control and media commands delivered via the RESTful API as described in this guide. The lighttpd server includes a Fast Common Gateway Interface (FCGI) process, which allows efficient interfacing between PowerMedia XMS processes.

The PowerMedia XMS translates RESTful commands into the PowerMedia HMP media engine's low-level API. The media engine itself handles SIP calls, plays/records multimedia, and mixes multimedia conferences.

PowerMedia XMS provides two call control models:

- First party call control (1PCC)
  The application sends commands to the PowerMedia XMS to establish SIP calls on the application's behalf. In this model, the application does not need to be involved in making or receiving SIP calls and related SDP negotiation.
- Third party call control (3PCC)
  The application handles SIP calls signaling and SDP negotiation, and the PowerMedia XMS only performs media processing operations.

RESTful API Description

The PowerMedia XMS RESTful API uses a Representational State Transfer (RESTful) web service. This web service is a software system designed to support interoperable machine-to-machine interactions over a network, using the HTTP protocol.

The RESTful API consists of a series of requests and responses built around the transfer of representations of "resources". These resources are accessed through Universal Resource Indicators (URIs).

RESTful client-server architecture is where clients initiate requests to servers and servers process the requests and return appropriate responses.

In a RESTful application, the http client is the application which contains the business logic and PowerMedia XMS is the http server which handles the client request and processes the media commands.

Client Side Technologies

The "client side" refers to the client that communicates with the PowerMedia XMS and directs the session with the caller. Essentially, any language or operating system may be used to build a client. The main requirement is that the client supports HTTP and XML.

Listed below are some possible examples of client-side development platforms that can be used to command PowerMedia XMS services. Comments are included on multithreading, which is important for the event handler.

Java – This object-oriented, operating system-independent programming environment is fully multithreaded. Several XSD/XML parsers are available, as well as HTTP client class libraries. See the XML Schema Definition section for information on XSDs.

Note: The Dialogic Verification Demo used with the PowerMedia XMS is a Java application. Refer to the *Dialogic® PowerMedia™ XMS Quick Start Guide* for information about the Demo.

Python – This operating system-independent interpreted scripting language has POSIX threading available. HTTP protocol client library and Python XML/Schema processing tool are also available.

.NET – This Integral Microsoft Windows component supports the building and running of applications and XML web services. HTTP module and XSD schema definition tools are available.

Ruby – This open source scripting language contains a multiprocessing model that may be needed for the event handler. An HTTP client API and XSD validation tools are available.

C/C++ – These general purpose programming languages are fully multithreaded. cURL library (http://curl.haxx.se) is used for HTTP processing and Xerces C++ XML parser (http://xerces.apache.org/xerces-c) is used for XML. For a proof of concept, see http://www.dialogic.com/support/helpweb/helpweb.aspx/3584/powermedia_xms_restful_C_Sharp_demo/PM_XMS.

RESTful API with HTTP Methods

In the RESTful API, the four HTTP methods are translated to the actions shown in the following table.

HTTP Method	Request	Response
POST	Create a new resource	Contents of a newly created resource
PUT	Modify an existing resource	Contents of an updated resource
GET	Retrieve information for all instances of a specific resource type, or information regarding a specific resource	Contents of resource information
DELETE	Delete an existing resource	N/A

RESTful API Request/Response Model

The HTTP request/response model is the mechanism by which media control functionality is invoked. A RESTful HTTP request is sent to the PowerMedia XMS. The HTTP response carries the resulting response code of the operation, as well as a response body if it applies to the specific operation. The payload type used for the message body is XML.

All Call Resources

If a client wished to retrieve a list of all call resources currently active on the PowerMedia XMS, it would issue an HTTP **GET** request. The HTTP **GET** request would be sent on the web service with the IP address of the PowerMedia XMS <server>. For example:

```
http://<server>/default/calls?appid=app
```

If successful, the response code to the HTTP **GET** would be `200 OK`. The response body would resemble the following sample:

```
<web_service version="1.0">
   <calls_response size="2">
   <call_response  appid="master"
    identifier="123zdasdkz"
    href=http://<server>/default/calls/123zdasdkz
    cpa = "yes"
    signaling = "yes"
    source_uri=sip:frank@10.20.34.3
    call_type="inbound" />
   <call_response
    identifier="178zdasdkz"
    href=http://<server>/default/calls/178zdasdkz
    cpa = "no"
    signaling = "no"
    sdp=[sdp]
    call_type="3pcc" />
  </calls_response>
</web_service>
```

The above sample shows a client requesting information for all calls with a response of two active identifiers along with the attributes of each call resource.

Single Call Resource

If a client wanted to retrieve information for only a single specific call resource, it would invoke the following HTTP **GET** request. The specific call identifier is part of the GET URL.

```
http://<server>/default/calls/1279697438?appid=app
```

If successful, the response code to the HTTP **GET** would be `200 OK`. The response body would be as follows:

```
<web_service version="1.0">
   <call_response  appid="master"
    identifier="1279697438"
    href=http://<server>/default/calls/1279697438
    cpa = "yes"
    signaling = "yes"
    source_uri=sip:frank@10.20.34.3
    call_type="inbound">
   </calls_response>
</web_service>
```

Additional request/response examples are contained within Resource-Based Components.

All XML sent to the PowerMedia XMS should have proper XML escape codes within string content. For example, when a <call> contains SDP info in the sdp="" attribute, newlines need to be converted to the proper XML escape code "
" and "&#xD".

```
<call sdp="v=0&#xA;o=sipclient 1376422095 1376422096 IN IP4
10.20.129.100&#xA;s=sipclient&#xA;c=IN IP4 10.20.129.100&#xA;t=0
0&#xA;m=audio 49162 RTP/AVP &#xA;a=rtpmap:0 pcmu/8000&#xA;a=sendrecv&#xA;"
media="audiovideo" signaling="no"/>
```

XML Schema Definition

PowerMedia XMS uses an XML schema definition (also referred to herein as "XSD"). The XSD formally describes the structure, content, and semantics of the XML payload for the PowerMedia XMS RESTful API call and media commands.

An XSD may be used to generate client-side code, allowing contents of XML documents to be treated as objects. The generated code usually enforces type-checking, thus supporting client-side validation of the XML payload before it is sent to the PowerMedia XMS.

Definitions of individual elements are referenced throughout this guide. The full XSD is provided in XML Schema Definition of Elements.

PowerMedia XMS RESTful API is designed using the following XML Schema declarations:

Element
An element describes the data it contains. It consists of a name and data type. When an element definition contains additional elements or attributes, it is a complex type.

```
               <xs:element name="call_response">
```

Attribute
An attribute is a simple type definition that cannot contain other elements. Attribute names are always within quotation marks.

```
               <xs:attribute name="media">
```

Sequence
Specifies the order in which attributes or elements within a complex type must be listed.

```
               <xs:element name="call_response">
                       <xs:complexType>
                     <xs:sequence>
                           <xs:element ref="call_action" minOccurs="0" />
                     </xs:sequence>
                     <xs:attribute name="signaling" type="boolean_type" />
                     <xs:attribute name="media" type="media_type" />
```

Complex Type
Defines an element containing other elements and attributes or mixed content (elements and text).

```
<xs:element name="call_response">
       <xs:complexType>
              <xs:sequence>
                    <xs:element ref="call_action" minOccurs="0" />
              </xs:sequence>
              <xs:attribute name="signaling" type="boolean_type" />
              <xs:attribute name="destination_uri" type="xs:string" />
              <xs:attribute name="source_uri" type="xs:string" />
              <xs:attribute name="call_type" type="call_type_option" />
              <xs:attribute name="sdp" type="xs:string"/>
              <xs:attribute name="cpa" type="boolean_type" />
              <xs:attribute name="media" type="media_type" />
              <xs:attribute name="dtmf_mode" type="dtmf_mode_option" />
              <xs:attribute name="async_dtmf" type="boolean_type" />
              <xs:attribute name="async_tone" type="boolean_type" />
              <xs:attribute name="cleardigits" type="boolean_type" />
              <xs:attributeGroup ref="response_attrgroup" />
       </xs:complexType>
</xs:element>
```

Simple Type
Creates a constrained data type for an element or attribute value.

```
<xs:simpleType name="call_type_option">
       <xs:restriction base="xs:string">
              <xs:enumeration value="inbound" />
              <xs:enumeration value="outbound" />
              <xs:enumeration value="3pcc" />
       </xs:restriction>
</xs:simpleType>
```

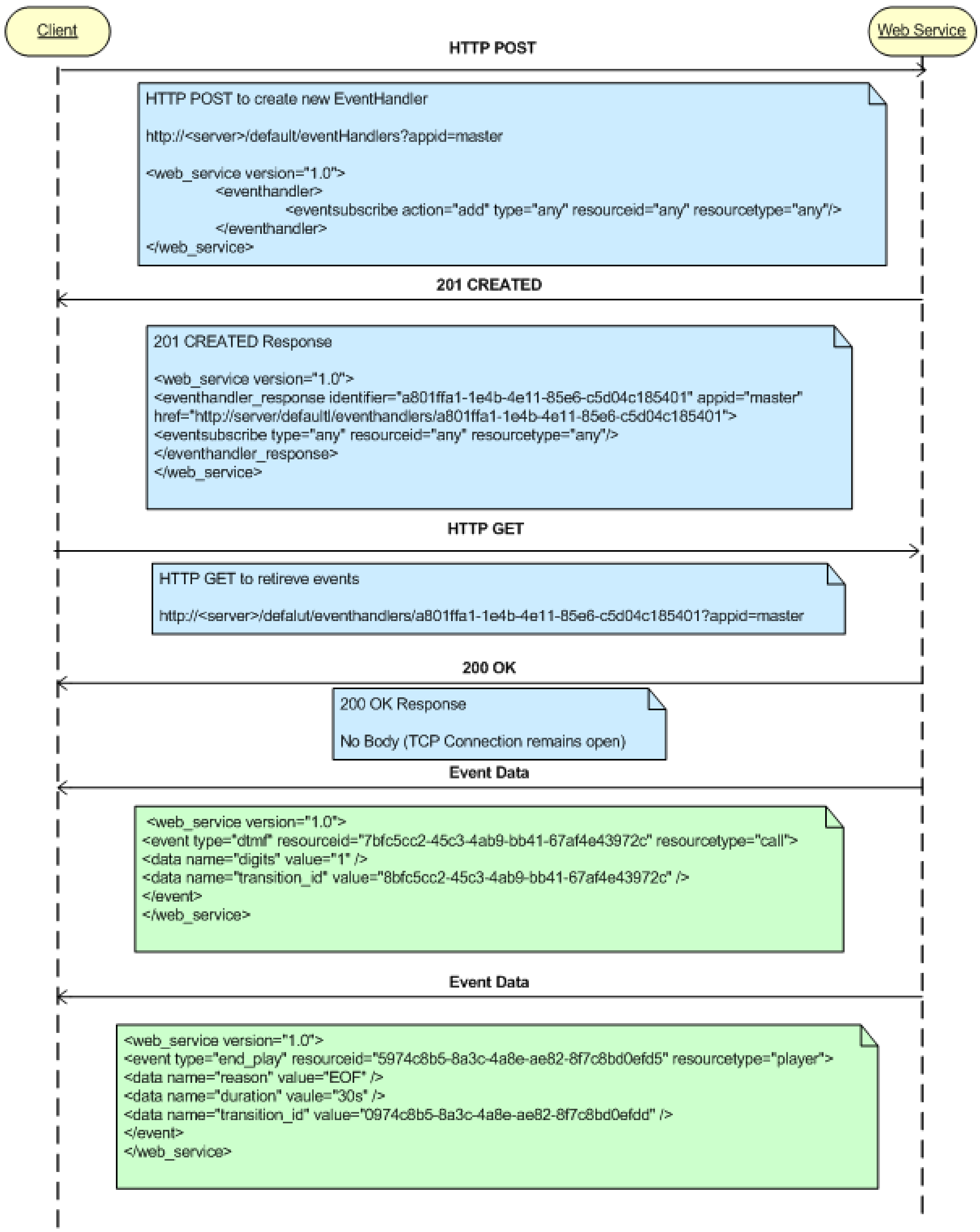
Refer to the specific resource-based element sections for more information.

Event Streaming

While most RESTful applications fit well into the HTTP request/response model, telephony applications must be able to handle unsolicited events such as digit detection and play completion. This concept is called Comet or HTTP event streaming. In a normal HTTP interaction, the client sends a request to the server, which processes it and sends the HTTP response. The connection between the client and server is then closed. This process will take place continuously as long as the web service is running; however, with HTTP event/data streaming, a reliable TCP connection remains open after the response is sent from the server, allowing the server to continue to send raw data to the client without solicitation or without client request.

For PowerMedia XMS, HTTP event streaming is implemented in the eventhandler resource. When the client wishes to receive asynchronous events, it uses an HTTP **POST** to create an eventhandler and subscribe to specific event types. The client then performs an HTTP **GET** on the newly created eventhandler and the PowerMedia XMS RESTful API responds with a `200 OK`; however, the TCP connection remains open. Any event data related to resources and event types are pushed to the client until the Eventhandler is deleted by the client.

The following diagram provides an example scenario where a client creates an eventhandler and receives digit detection and play completion events:



3. Resource-Based Element Introduction

There are four (4) resource-based elements used by the PowerMedia XMS RESTful web service.

- Call Resource
- Conference Resource
- Event Handler Resource
- MRCP Resource

These elements are used in conjunction with one another to direct the PowerMedia XMS to make and receive calls, handle media during a call, manipulate audio and video conferences, invoke ASR/TTS speech services, and to catch events relating to calls, conferences and their media.

Each element makes use of the various HTTP methods – POST, PUT, DELETE and GET. The resource-based element chapters in this document contain HTTP method tables that define the request body content type if a request body is allowed. In addition, the tables supply the possible return code values as well as the response body content. The tables also contain a sample payload for that specific resource-based element type. This XML content is used in both HTTP requests and responses. Refer to the specific resource-based element sections for information about each of the elements.

Applications and Application IDs (appid)

The *appid* shown in the URL request examples is for identifying the resources used, owned and created by a specific application. For example:

```
<web_service version="1.0">
  <call_response
    appid="[appid]"
    identifier="[call_id]"
    href="[base_url]/calls/[call_id]?appid=[appid]"
    signaling = "yes"
    source_uri=[uri]
    sdp=[sdp]
    call_type="inbound" />
</web_service>
```

Discrete appids are defined so that multiple applications may be simultaneously run on a single PowerMedia XMS. The appid indicates the ownership of a RESTful resource so that each resource that is created has an associated appid. The resources can only be viewed, modified, or deleted by an application with a matching appid. The appid is used throughout the PowerMedia XMS RESTful API to identify the intended application.

Note: The appid is pre-defined on the Routing page of the PowerMedia XMS Admin Console (also referred to herein as "Console"), which is used for post-operating system installation and configuration tasks. New appids may be added, or unwanted appids removed on the Routing page. Refer to the *Dialogic® PowerMedia™ XMS Installation and Configuration Guide* for detailed information about the Console.

Time Values

Values that represent time in the RESTful API are specified in whole numbers of seconds ("s") or milliseconds ("ms") whichever is appropriate.

For example:

```
<send_dtmf digits="1234" duration="100ms" interval="100ms" level="-10dB"/>
<play offset="2s" repeat="3" delay="0s" terminate_digits="#" max_time="infinite"
skip_interval="10s">
```

4. API Resources

The *appid* shown in the following URL request is for identifying the resources used, owned, and created by the application. All Resource Base URI:

```
http://[ipaddress:port]/default/
default port: 81
```

Note: "/default/" is a placeholder.

The resources provided by the RESTful API will differ between the various modes.

```
default appid=app
```

The table below lists all available resources, their sub-resources, valid HTTP methods that may be used with them and attributes for which they are valid.

Clicking on the resource or sub-resource will:

- Provide its definition
- Provide valid values for the parameters that can be set
- Define how each valid method affects the resource
- Give an example of a request and a response payload

Resource	Sub-Resource	HTTP Methods Supported
Call Resource /calls?appid=[app_id]		GET, POST
	Call Sub-Resource /calls/[call_id]?appid=[app_id]	GET, PUT, DELETE
Conference Resource /conferences?appid=[app_id]		GET, POST
	Conference Sub-Resource /conferences/[conference_id]?appid=[app_id]	GET, PUT, DELETE
MRCP Resource /mrcps?appid=[app_id]		GET, POST
	MRCP Sub-Resource /mrcps/[mrcp_id]?appid=[app_id]	GET, PUT, DELETE
Event Handler Resource /eventhandlers?appid=[app_id]		GET, POST
	Event Handler Sub-Resource /eventhandlers/[eventhandler_id]?appid=[app_id]	GET, PUT, DELETE

5. List of Available Resources

Call Resource

The Call Resource creates and manages the media/signal connection between the remote media endpoint (typically a SIP endpoint) and the PowerMedia XMS.

The Call Resource has the following types:

- **Inbound**
  This call resource is created by the PowerMedia XMS RESTful server when an incoming call is received. The application is then informed of the inbound call via the eventhandler resource.
- **Outbound**
  This call resource is created by an application that wishes to make a media stream connection from the PowerMedia XMS RESTful server to a SIP endpoint.
- **3PCC**
  This call resource is requested by the application without requesting the PowerMedia XMS RESTful server to provide signaling control. The call resource will be created based on the Session Description Protocol (SDP) info that is provided by the application. PowerMedia XMS can handle SDP from standard SIP endpoints. 3PCC outbound calls should have SDP="" so that PowerMedia XMS can generate the SDP for the app to include in the INVITE while 3PCC incoming calls should have SDP set to whatever is in the incoming INVITE.

  Note: In order to set the PowerMedia XMS to 3PCC mode, you need to set the signaling="no".

Media-related properties and actions associated with the media connection are defined in this section. These include play, playcollect, playrecord, overlay, join/unjoin, and stop.

For details on call sub-resources, see the Call Sub-Resource section.

The following tables show the HTTP methods that can be used with one or more calls.

Note: The payloads shown are examples only as there are many possible variations.

calls

Resource URI

`/calls?appid=[app_id]`

HTTP GET

Retrieves all available call resources.

`GET /calls?appid=[app_id]`

Response Payload Example


```
<web_service version="1.0">

   <calls_response size="2">

         <call_response appid="app" async_dtmf="yes" async_tone="yes"
audio="sendrecv" call_type="inbound"
```

```
                cleardigits="no" connected="yes" cpa="no"
destination_uri="sip:sip@10.20.129.100" dtmf_mode="rfc2833"
                href="http://10.20.129.100:81/default/calls/8ae87129-b334-4d8a-bec6-
5d4ddeba5649" .
                identifier="8ae87129-b334-4d8a-bec6-5d4ddeba5649"
info_ack_mode="automatic" media="audiovideo" signaling="yes"
                source_uri="sip:Username@10.20.129.113:5060" video="sendrecv">
         </call_response>
         <call_response appid="app" async_dtmf="yes" async_tone="yes"
audio="sendrecv" call_type="inbound"
                cleardigits="no" connected="yes" cpa="no"
destination_uri="sip:sip@10.20.129.100" dtmf_mode="rfc2833"
                href="http://10.20.129.100:81/default/calls/f1e9040f-4ca2-4dd6-81e6-
c665385ffde8"
                identifier="f1e9040f-4ca2-4dd6-81e6-c665385ffde8"
info_ack_mode="automatic" media="audiovideo" signaling="yes"
                source_uri="sip:Username@10.20.129.113:5060" video="sendrecv">
         </call_response>
   </calls_response>
</web_service>
```


HTTP POST

Creates a call resource.

```
POST /calls?appid=[app_id]
```

Create Call Types

- Outbound
- 3PCC (application handles SIP call signaling)

Request Payload Attributes

Parameter	Default	Optional	Description
signaling	"yes"	*	Specifies if signaling is done by this media server ("yes") or a third party application server ("no").

Parameter	Default	Optional	Description
sdp	Set by system	*	Session Description Protocol data. Only used for 3rd party call control (3PCC). **Note:** If the data contains newlines or carriage returns, make sure that they are replaced with the XML equivalent of "&#xA" prior to sending.
media	"audio"	*	Sets the media type supported by the call. Values: • "audio" • "audiovideo"
dtmf_mode	"rfc2833"	*	Specifies the signaling mode for DTMF digits. Values: • "inband" • "outofband" • "rfc2833"
cpa	"no"	*	Specifies if call progress detection is used for an outbound call (signaling only). Values: • "yes" • "no"

Parameter	Default	Optional	Description
info_ack_mode	"automatic"	*	Specifies how INFO events are acknowledged (signaling only). Values: • "automatic" • "manual"
rx_volume	(none)	*	Volume adjustments are allowed between +31dB and -32dB. Both absolute (default) and relative adjustments are supported. Values: • "+3dB;relative" • "+3dB;absolute"
tx_volume	(none)	*	Volume adjustments are allowed between +31dB and -32dB. Both absolute (default) and relative adjustments are supported. Values: • "+3dB;relative" • "+3dB;absolute"

Parameter	Default	Optional	Description
async_dtmf	(none)	*	Specifies if DTMF digits should be reported as events instead of being buffered internally (default). When active, the application must ensure that when using the any of the play APIs, any unused digit processing parameters are cleared. This is to avoid digits being processed both internally and by the application. Values: • "yes" • "no"
cleardigits	(none)	*	The parameter is considered when async_dtmf is set to "yes" and specifies whether previous, buffered, input should be discarded. Values: • "yes" • "no"
async_tone	(none)	*	Specifies if tones are reported as events outside of a play_collect action. Values: • "yes" • "no"
destination_uri	(none)	*	Destination address. For SIP, this is the Request-URI.
source_uri	(none)	*	Caller address. For SIP, this is the From header.

Parameter	Default	Optional	Description
called_uri	(none)	*	Logical destination address. For SIP, this is the To header.
display_name	(none)	*	Caller's display name.
dial_timeout	"30s"	*	Maximum time to wait for the call to be answered by the called party.
encryption	"none"	*	Media stream (RTP) encryption. Values: • "none" • "dtls"
ice	"no"	*	Use ICE (Interactive Connectivity Establishment) to configure media streams (RTP). Values: • "no" • "yes"

Request Payload Example

Outbound

```
<web_service version="1.0">
  <call  media="audiovideo" signaling="yes" dtmf_mode="rfc2833" async_dtmf="yes"
async_tone="yes" rx_delta="+0dB"
         tx_delta="+0dB" destination_uri="sip:xmstool@10.20.129.102"
source_uri="sip:xmstool@146.152.124.182" cpa="no" />
</web_service>
```

3PCC

```
<web_service version="1.0">
  <call sdp="[sdpinfo]" media="audiovideo" signaling="no"/>
</web_service>
```

Response Payload Example

Outbound


```
<web_service version="1.0">
     <call_response identifier="7f0e358f-5786-41df-bad3-866f73d044a7" appid="app"
               href="http://10.20.129.100:81/default/calls/7f0e358f-5786-41df-bad3-866f73d044a7"
               connected="no" signaling="yes" cpa="no" call_type="outbound"
               media="audiovideo"
               dtmf_mode="rfc2833"
               destination_uri="sip:xmstool@10.20.129.102"
               source_uri="sip:xmstool@146.152.124.182"
               async_dtmf="yes" async_tone="yes" cleardigits="no" info_ack_mode="automatic">
     </call_response>
</web_service>
```


3PCC


```
<web_service version="1.0">
     <call_response identifier="ca494fe7-17d7-4b6a-a7d0-e89592eef262" appid="app"
            href="http://10.20.129.61:81/default/calls/ca494fe7-17d7-4b6a-a7d0-e89592eef262"
            sdp="v=0&#xA;o=xmserver 1376426725 1376426726 IN IP4 10.20.129.61&#xA;s=xmserver&#xA;c=IN
            IP4 10.20.129.61&#xA;t=0 0&#xA;m=audio 49152 RTP/AVP 9 0 8 96 97 4 18 98
            101&#xA;a=rtpmap:9 g722/8000&#xA;a=rtpmap:0 pcmu/8000&#xA;a=rtpmap:8
            pcma/8000&#xA;a=rtpmap:96 g726-32/8000&#xA;a=rtpmap:97 amr/8000&#xA;a=fmtp:97 octet-
            align=0&#xA;a=rtpmap:4 g723/8000&#xA;a=fmtp:4 annexa=yes&#xA;a=rtpmap:18
            g729/8000&#xA;a=fmtp:18 annexb=no&#xA;a=rtpmap:98 amr-wb/16000&#xA;a=fmtp:98 octet-
            align=0&#xA;a=rtpmap:101 telephone-event/8000&#xA;a=fmtp:101 0-
            15&#xA;a=sendrecv&#xA;m=video 57344 RTP/AVP 100 98 34 96 97
            102&#xA;b=AS:1000&#xA;a=rtpmap:100 h264/90000&#xA;a=fmtp:100 profile-level-id=42001F;
            packetization-mode=1; max-br=768&#xA;a=rtpmap:98 mp4v-es/90000&#xA;a=fmtp:98 profile-
            level-id=3 MaxBR=3840&#xA;a=rtpmap:34 h263/90000&#xA;a=fmtp:34 CIF=1; CIF=2; CIF=3;
            QCIF=1; QCIF=2&#xA;a=rtpmap:96 h263-1998/90000&#xA;a=fmtp:96 CIF=1; CIF=2; CIF=3; QCIF=1;
            QCIF=2&#xA;a=rtpmap:97 h263-2000/90000&#xA;a=fmtp:97 CIF=1; CIF=2; CIF=3; QCIF=1;
```

```
            QCIF=2&#xA;a=rtpmap:102 vp8/90000&#xA;a=fmtp:102 max-fr=30; max-
            fs=1200&#xA;a=sendrecv&#xA;"
            signaling="no" cpa="no" call_type="3pcc"
            media="audiovideo"
            dtmf_mode="rfc2833"
            source_uri="sip:xmstool@146.152.124.182"
            async_dtmf="yes" async_tone="yes" cleardigits="no" encryption="none"
ice="no" info_ack_mode="automatic">
      </call_response>
</web_service>
```

Note: If you send a request in 3PCC mode with an empty SDP="", PowerMedia XMS will allocate an internal device and return back the SDP. This allows SDP to initiate and outbound call with early media.

Call Concepts

This section contains a higher-level look at various aspects of PowerMedia XMS call behavior.

Asynchronous Tones

- Audio tones can be used to both terminate operations and can also be delivered to the application as asynchronous events.
- To terminate a playrecord or play operation, set terminate_digits to the desired DTMF digit value (0-9, * and #). If the operation is terminated this way, the end_playrecord or endplay event will reference the digit collected to end the operation.
- To have an async DTMF event delivered to your application outside of its use as a play/record termination, async_dtmf=yes should be set for a call resource.
- User-defined tone detection is set up using tone templates, which are created in the PowerMedia XMS GUI in the Tones screen (see the *Dialogic® PowerMedia™ XMS Installation and Configuration Guide* for more information). Set async_tones=yes for a call resource and detection of the defined tones is activated.
- DTMF events are delivered as event type "dtmf", with a name of "digits" and a value corresponding to the digit collected.
- Async tone events are delivered as event type "tone", with a name of "tone" and a value corresponding to the name assigned when the user-defined tone was created.

Playcollect and User-Defined Tones

A playcollect operation usually is used to collect DTMF tones but may collect user-defined tone as well.

To do this, the attribute tone_detection=yes must be set when the playcollect is initiated. The end_playcollect event will reference the user-defined name of the tone collected and the reason parameter will be set to tone. Duration (in milliseconds) refers to the length of the play operation, not the duration of the tone.

Early Media

Early media refers to the ability to play media (audio and/or video) on an inbound call to the caller before a call is answered. The call must first be Accepted before the media play operation can be started. To fully complete the connection, the call must be answered. A PUT should be used for these messages.

Media File Locations

The default location for media files used by the PowerMedia XMS RESTful API is displayed when the Media menu is chosen on the PowerMedia XMS Console. See the Media section in the *Dialogic® PowerMedia™ XMS Installation and Configuration Guide*.

Files for a given RESTful application are usually grouped under a directory reflecting the application name. Directories/applications delivered with the system are "verification" (for the RESTful Verification Demo) and "xmstool", for the XMSTool RESTful Utility.

Thus, the file:// URL used in a RESTful Play command would point at a set of audio and video files named *xmstool_play.wav* and *xmstool_play.vid*, both located in the xmstool directory in the default media location:

```
<play_source location="file://xmstool/xmstool_play"/>
```

Similarly, the location of a file for a recording is given in the same way:

```
<playrecord recording_uri="file://xmstool/xmstool_recording" max_time="10s" offset="0s"
repeat="0" delay="1s"
```

Call Sub-Resource

For details on call resources, see the Call Resource section.

call

Resource URI

```
/calls/[call_id]?appid=[app_id]
```

HTTP GET

Retrieves an available call resource.

```
GET /calls/[call_id]?appid=[app_id]
```

Response Payload

```
<web_service version="1.0">
    <call_response appid="app" async_dtmf="yes" async_tone="yes" audio="sendrecv"
call_type="inbound"
              cleardigits="no" connected="yes" cpa="no"
destination_uri="sip:sip@10.20.129.100" dtmf_mode="rfc2833"
              href="http://10.20.129.100:81/default/calls/8ae87129-b334-4d8a-bec6-
5d4ddeba5649" .
              identifier="8ae87129-b334-4d8a-bec6-5d4ddeba5649"
info_ack_mode="automatic" media="audiovideo" signaling="yes"
              source_uri="sip:Username@10.20.129.113:5060" video="sendrecv">
    </call_response>
</web_service>
```

HTTP PUT

Updates a call resource.

`PUT /calls/[call_id]?appid=[app_id]`

Accept/Answer Incoming Call

Request Payload Attributes

Parameter	Default	Optional	Description
answer	(none)	*	Answer an incoming call. Values: • "yes" • "no"
accept	(none)	*	Accept an incoming call. Values: • "yes" • "no"
media	"audio"	*	Sets the media supported by the call. Values: • "audio" • "audiovideo" This parameter has no effect if the call has been accepted already.
dtmf_mode	"rfc2833"	*	Specifies the signaling mode for DTMF digits. Values: • "inband" • "outofband" • "rfc2833"

Parameter	Default	Optional	Description
early_media	"no"	*	Enable early media. Values: • "yes" • "no" This parameter has no effect if answer=yes.
info_ack_mode	"automatic"	*	Specifies how INFO events are acknowledged. Values: • "automatic" • "manual"
async_completion	"no"	*	Values: • "yes" the api will return immediately and completion is indicated by an ANSWERED event. • "no" the api will block until the call has been fully answered. This parameter has effect if answer=yes.
encryption	(none)	*	Media stream (RTP) encryption. Values: • "none" • "dtls"

Parameter	Default	Optional	Description
ice	(none)	*	Use ICE (Interactive Connectivity Establishment) to configure media streams (RTP). Values: • "no" • "yes"
content_type	(none)	*	Mime type describing content (answer only).
content	(none)	*	Data content (answer only).

Request Payload Example

accept

```
<web_service version="1.0">
  <call accept="yes" early_media="yes" />
</web_service>
```

answer

```
<web_service version="1.0">
  <call answer="yes" async_completion="yes"/>
</web_service>
```

Response Payload Example

accept

```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
            href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
            connected="yes" signaling="yes" cpa="no" call_type="inbound"
            media="audio"
            dtmf_mode="rfc2833"
            source_uri="sip:Username@10.20.129.113:5060"
            async_dtmf="no" async_tone="no" cleardigits="no"
info_ack_mode="automatic" early_media="yes">
     </call_response>
</web_service>
```

answer


```
<web_service version="1.0">
      <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
             href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-52f57e00314b"
             connected="yes" signaling="yes" cpa="no" call_type="inbound"
             media="unknown"
             dtmf_mode="rfc2833"
             source_uri="sip:Username@10.20.129.113:5060"
             async_dtmf="no" async_tone="no" cleardigits="no"
info_ack_mode="automatic" early_media="yes">
      </call_response>
</web_service>
```


Modify Call Attributes

Request Payload Attributes

Parameter	Default	Optional	Description
sdp	(none)	*	Only used in 3rd party call control (3PCC).
rx_delta	(none)	*	Volume adjustments are allowed between +31dB and -32dB. Both absolute (default) and relative adjustments are supported. Values: • "+3dB;relative" • "+3dB;absolute"
tx_delta	(none)	*	Volume adjustments are allowed between +31dB and -32dB. Both absolute (default) and relative adjustments are supported. Values: • "+3dB;relative" • "+3dB;absolute"

Parameter	Default	Optional	Description
async_dtmf	(none)	*	Specifies if DTMF digits should be reported as events instead of being buffered internally (default). When active, the application must ensure that when using the any of the play APIs, any unused digit processing parameters are cleared. This is to avoid digits being processed both internally and by the application. Values: • "yes" • "no"
cleardigits	(none)	*	The parameter is considered when async_dtmf is set to "yes" and specifies whether previous, buffered, input should be discarded. Values: • "yes" • "no"
async_tone	(none)	*	Specifies if tones are reported as events outside of a play_collect action. Values: • "yes" • "no"

Request Payload Example

```
<web_service version="1.0">
  <call async_dtmf="yes" async_tone="yes" rx_delta="+3dB" tx_delta="+3dB" />
</web_service>
```

Response Payload Example


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
            href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
            connected="yes" signaling="yes" cpa="no" call_type="inbound"
            media="audio"
            dtmf_mode="rfc2833"
            source_uri="sip:Username@10.20.129.113:5060"
            async_dtmf="yes" async_tone="yes" cleardigits="no"
info_ack_mode="automatic" early_media="yes">
     </call_response>
</web_service>
```


Perform Call Action

- play
- update_play
- record
- playcollect
- playrecord
- overlay
- stop
- join/unjoin
- add_party/update_party
- remove_party
- send_dtmf
- send_info/send_info_ack
- transfer
- redirect
- hangup
- get_call_info

play_source_attributes

Parameter	Default	Optional	Description
location	(none)		The URI of the media to be played e.g. "file://...", "rtsp://...", " image:", "http(s)://..." DEPRECATED use audio_uri and video_uri.
base_audio_uri	(none)	*	Base URI prefix for audio_uri URIs e.g. "file://...", "rtsp://...", "http(s)://..."
audio_uri	(none)		The URI of the audio media to be played e.g. "file://...", "rtsp://...", "http(s)://..." or <filename>.<ext> if the base_audio_uri has been set. Multiple URIs can be specified by using the newline character '\n' as a separator. Only URIs of type "file://..." are currently supported when playing multiple files.
audio_type	(none)		The mime-type of the audio media. • "audio/x-wav" • "audio/basic" • "audio/x-alaw-basic" • "audio/G723" • "audio/G726" • "audio/G729"
base_video_uri	(none)	*	Base URI prefix for video_uri URIs e.g. "file://...", "rtsp://...", "http(s)://..."

Parameter	Default	Optional	Description
video_uri	(none)	*	The URI of the video media to be played e.g. "file://...", "rtsp://...", "http(s)://...or <filename>.<ext> if the base_video_uri has been set. Multiple URIs can be specified by using the newline character '\n' as a separator. Only URIs of type "file://..." are currently supported when playing multiple files.
video_type	(none)	*	The mime-type of the video media. • "video/x-vid" • "image/jpeg"

recording_audio_mime_params

Parameter	Default	Optional	Description
codec	(none)	*	Sets the audio codec. Values: • "L16" • "mulaw" • "alaw"
rate	(none)	*	Sets the audio rate. Values: • "8000" • "16000 (L16 only)"

recording_video_mime_params

Parameter	Default	Optional	Description
codec	(none)	*	Sets the video codec. Values: • "h263" • "h264" • "mp4v-es"
profile	(none)	*	Sets the video profile.
level	(none)	*	Sets the video level.
framerate	(none)	*	Sets the video framerate.
maxbitrate	(none)	*	Sets the video maxbitrate.
height	(none)	*	Sets the video height.
width	(none)	*	Sets the video width.

dvr_setting_attributes

Parameter	Default	Optional	Description
forward_key	"1"	*	Defines the DTMF key [0-9,*,#] used to skip forwards.
backward_key	"2"	*	Defines the DTMF key [0-9,*,#] used to skip backwards.
pause_key	"3"	*	Defines the DTMF key [0-9,*,#] used to pause playback.
resume_key	"4"	*	Defines the DTMF key [0-9,*,#] used to resume playback.
restart_key	"5"	*	Defines the DTMF key [0-9,*,#] used to restart playback.

param_attributes

Parameter	Default	Optional	Description
name	(none)		Sets parameter name.
value	(none)		Sets parameter value.

sip_headers_attributes

Parameter	Default	Optional	Description
raw_sip_headers	(none)		Raw SIP headers, delimited by the <CR><LF> end-of-line characters.
params	(none)		Refer to param_attributes and get_call_info for examples.

play

Request Payload Attributes

Parameter	Default	Optional	Description
play_source	(none)		Refer to play_source_attributes and get_call_info for examples.
dvr_setting	(none)	*	Refer to dvr_setting_attributes.
offset	"0"	*	Specifies the time offset from where the play should start. Note, the 'offset' is applied to the initial play only.
repeat	"0"	*	Number of times to repeat the play. Use "infinite" to repeat indefinitely. "file://" URIs only.

Parameter	Default	Optional	Description
delay	"1s"	*	Time delay between repeated plays.
terminate_digits	"#"	*	The digit or digits [0-9,*,#] used to terminate the play (call only).
max_time	"infinite"	*	Limit the playback time to this value.
skip_interval	"1s"	*	Defines the amount of time to skip on the 'forward' and 'backwards' actions (call only).
no_cache		*	Cache-control for http URIs. Values: • "yes" does not use caching. • "no" uses caching. The default is set by the configuration.
max_age		*	Cache-control for http URIs, the maximum age, in seconds, of a cached file. The default is set by the configuration.
max_stale		*	Cache-control for http URIs, the number of seconds that a cached file may exceed its expiration time by and still be a considered as fresh. The default is set by the configuration.

Parameter	Default	Optional	Description
fetch_timeout	"300s"	*	http URIs, the maximum time in seconds to retrieve the file. This is the overall period of the transaction.

Request Payload Example

```
<web_service version="1.0">
  <call>
   <call_action>
      <play offset="0s" repeat="0" delay="1s" terminate_digits="#"
skip_interval="1s">
         <play_source audio_uri="file://verification/play_menu.wav"
audio_type="audio/x-wav" />
         <dvr_setting forward_key="1" backward_key="2" pause_key="3" resume_key="4"
restart_key="5"/>
      </play>
   </call_action>
</call>
</web_service>
```

Response Payload Example

```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
              <play transaction_id="9d608231-a164-4102-b9e6-3ba1f0671a53"
                    max_time="infinite"
                    fetch_timeout="300s"
                    offset="0s"
                    delay="1s"
                    repeat="0"
```

```
                    skip_interval="1s"
                    terminate_digits="#">
                    <play_source audio_uri="file://verification/play_menu.wav"
audio_type="audio/x-wav" />
                    <dvr_setting forward_key="1"
                                 backward_key="2"
                                 pause_key="3"
                                 resume_key="4"
                                 restart_key="5"/>
              </play>
           </call_action>
     </call_response>
</web_service>
```

update_play

Request Payload Attributes

Parameter	Default	Optional	Description
transaction_id	(none)		Media identifier, returned by play.
dvr_action	(none)		Values: • "backward" - skip backwards. • "forward" - skip forward. • "pause" - pause playback. • "restart" - jump back to the start. • "resume" - resume paused playback.

Request Payload Example


```
<web_service version="1.0">
  <call>
     <call_action>
        <update_play transaction_id="9d608231-a164-4102-b9e6-3ba1f0671a53"
dvr_action="pause"/>
     </call_action>
  </call>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
               <update_play transaction_id="9d608231-a164-4102-b9e6-3ba1f0671a53"
dvr_action="pause"/>
          </call_action>
     </call_response>
</web_service>
```


record

Request Payload Attributes

Parameter	Default	Optional	Description
recording_audio_mime_params	(none)		Refer to recording_audio_mime_params.
recording_video_mime_params	(none)		Refer to recording_video_mime_params.
terminate_digits	"#"	*	The digit or digits [0-9,*,#] used to terminate the play_record.

Parameter	Default	Optional	Description
recording_uri	(none)		A filename "file://..." which must refer to an existing directory or "http://" ("https://"). See also media DEPRECATED.
recording_audio_uri	(none)		The URL of the audio media to be recorded e.g. "file://...", "http(s)://..."
recording_audio_type	(none)		The mime-type of the audio media. • "audio/x-wav" • "audio/basic" • "audio/x-alaw-basic" • "audio/G723" • "audio/G726" • "audio/G729"
recording_video_uri	(none)		The URL of the video media to be recorded e.g. "file://...", "http(s)://..."
recording_video_type	(none)		The mime-type of the video media. • "video/x-vid"
max_time	"infinite"	*	The maximum length of time to record. Use "infinite" for no limit.
max_silence	"infinite"	*	The maximum length of silence to record after audio has been detected. Use "infinite" for no limit.
noinput_timeout	"infinite"	*	The maximum time to wait for audio to be detected. Use "infinite" for no limit.

Request Payload Example


```
<web_service version="1.0">
  <call>
    <call_action>
       <record terminate_digits="#" max_time="10s"
           recording_audio_uri="file://verification/beta_recorded_file.wav"
           recording_audio_type="audio/x-wav"
           recording_video_uri="file://verification/beta_recorded_file.vid"
           recording_video_type="video/x-vid" >
           <recording_video_mime_params codec="h264" level="3.1" height="480"
width="640" maxbitrate="768000" framerate="15"/>
           <recording_audio_mime_params codec="mulaw" rate="8000"/>
       </record>
    </call_action>
  </call>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
     <call_response identifier="7b3652ed-594f-482d-a349-7625c7cf86ef" appid="app"
           href="http://10.20.129.61:81/default/calls/7b3652ed-594f-482d-a349-
7625c7cf86ef"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audiovideo"
           dtmf_mode="rfc2833"
           destination_uri="sip:sip@10.20.129.61"
           source_uri="sip:Username@10.20.129.115:5060"
           async_dtmf="yes" async_tone="yes" cleardigits="no" encryption="none"
ice="no" info_ack_mode="automatic" audio="sendrecv" video="sendrecv">
           <call_action>
                <record transaction_id="b3407431-801f-4a44-ba8b-fa8c9fddd653"
                        terminate_digits="#"
                        max_time="10s"
                        max_silence="infinite"
                        noinput_timeout="infinite"
                        recording_audio_uri="file://verification/beta_recorded_file.
wav"
                        recording_audio_type="audio/x-wav"
                        recording_video_type="video/x-vid"
                        recording_video_uri="file://verification/beta_recorded_file.
vid">
```

```
                        <recording_audio_mime_params codec="mulaw" rate="8000"/>
                        <recording_video_mime_params codec="h264" level="3.1"
framerate="15" maxbitrate="768000" height="480" width="640"/>
                </record>
            </call_action>
        </call_response>
</web_service>
```

playrecord

Request Payload Attributes

Parameter	Default	Optional	Description
play_source	(none)		Refer to play_source_attributes.
recording_audio_mime_params	(none)		Refer to recording_audio_mime_params.
recording_video_mime_params	(none)		Refer to recording_video_mime_params.
offset	"0"	*	Specifies the time offset from where the play should start. Note, the 'offset' is applied to the initial play only.
repeat	"0"	*	Number of times to repeat the play. "file://" URIs only.
delay	"1s"	*	Time delay between repeated plays.
terminate_digits	"#"	*	The digit or digits [0-9,*,#] used to terminate the play_record.
recording_uri	(none)		A filename "file://..." which must refer to an existing directory. See also media DEPRECATED.
recording_audio_uri	(none)		The URL of the audio media to be recorded e.g. "file://...", "http(s)://..."

Parameter	Default	Optional	Description
recording_audio_type	(none)		The mime-type of the audio media. • "audio/x-wav" • "audio/basic" • "audio/x-alaw-basic"
recording_video_uri	(none)		The URL of the video media to be recorded e.g. "file://...", "http(s)://..."
recording_video_type	(none)		The mime-type of the video media. • "video/x-vid"
beep	"yes"	*	Play a tone before starting to record. Values: • "yes" • "no"
max_time	"infinite"	*	The maximum length of time to record. Use "infinite" for no limit.
max_silence	"infinite"	*	The maximum length of silence to record after audio has been detected. Use "infinite" for no limit. This feature is supported for calls only.
noinput_timeout	"infinite"	*	The maximum time to wait for audio to be detected. Use "infinite" for no limit. This feature is supported for calls only.

Parameter	Default	Optional	Description
barge	"yes"	*	Specifies whether dtmf digit input will barge the prompt and force transition to the record phase. Note that if the "barge" attribute is set to "no", the "cleardigits" attribute implicitly has the value "yes". Values: • "yes" • "no"
cleardigits	"no"	*	Specifies whether previous input should be considered or ignored for the purpose of barge-in. When it is set to "yes", any previously buffered digits are discarded. If it is set to "no", previously buffered digits will be considered. If "cleardigits" is set to "no" and "barge" is set to "yes", previously buffered digits will result in the recording phase starting immediately, and the prompt will not be played. Values: • "yes" • "no"
no_cache		*	Cache-control for http URIs. Values: • "yes" does not use caching. • "no" uses caching. The default is set by the configuration.
max_age		*	Cache-control for http URIs, the maximum age, in seconds, of a cached file. The default is set by the configuration.

Parameter	Default	Optional	Description
max_stale		*	Cache-control for http URIs, the number of seconds that a cached file may exceed its expiration time by and still be a considered as fresh. The default is set by the configuration.
fetch_timeout	"300s"	*	http URIs, the maximum time in seconds to retrieve or upload a file. This is the overall period of the transaction.

Request Payload Example


```
<web_service version="1.0">
  <call>
    <call_action>
      <playrecord recording_audio_uri="file://recorded_file.wav"
recording_audio_type="audio/x-wav"
                  max_time="10s" offset="0s" repeat="0" delay="1s"
terminate_digits="#"
                  beep="yes" barge="yes" cleardigits="yes" >
            <play_source audio_uri="file://verification/play_menu.wav"
audio_type="audio/x-wav"/>
       </playrecord>
    </call_action>
 </call>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
      <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
                     href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-
afee-52f57e00314b"
                     connected="yes" signaling="yes" cpa="no" call_type="inbound"
                     media="audio"
                     dtmf_mode="rfc2833"
                     source_uri="sip:Username@10.20.129.113:5060"
                     async_dtmf="no" async_tone="no" cleardigits="yes"
info_ack_mode="automatic" early_media="yes">
                     <call_action>
                         <playrecord transaction_id="a88aac4a-dd73-4a3f-8268-
792b90a5efb2"
```

```
                                        fetch_timeout="300s"
                                        terminate_digits="#"
                                        max_time="10s"
                                        beep="yes"
                                        barge="yes"
                                        cleardigits="yes"
                                        max_silence="infinite"
                                        noinput_timeout="infinite"
                                        recording_audio_uri="file://recorded_file.wav"
                                        recording_audio_type="audio/x-wav"
                                        offset="0s"
                                        delay="1s"
                                        repeat="0"
                                        terminate_digits="">
                                        <play_source
audio_uri="file://verification/play_menu.wav"
                                                    audio_type="audio/x-wav"/>
                         </playrecord>
                     </call_action>
       </call_response>
</web_service>
```

playcollect

Request Payload Attributes

Parameter	Default	Optional	Description
play_source	(none)		Refer to play_source_attributes.
offset	"0"	*	Specifies the time offset from where the play should start. Note, the 'offset' is applied to the initial play only.
repeat	"0"	*	Number of times to repeat the play. Use "infinite" to repeat indefinitely. "file://" URIs only.
delay	"1s"	*	Time delay between repeated plays.

Parameter	Default	Optional	Description
term_digits	"#"	*	The digit or digits [0-9,*,#] used to terminate the play_collect.
max_digits	Unlimited	*	The maximum number of digits to collect.
timeout	"infinite"	*	The maximum length of time to wait for the first digit or a tone. This time begins when the prompt phase ends.
interdigit_timeout	The value specified by the timeout parameter	*	The maximum length of time to wait for subsequent digits. This timeout is reset after each digit is received.
tone_detection	"no"	*	Enable tone detection. Values: • "yes" • "no"
barge	"yes"	*	Specifies whether dtmf digit input will barge the prompt and force transition to the collect phase. Note that if the "barge" attribute is set to "no", the "cleardigits" attribute implicitly has the value "yes". Values: • "yes" • "no"

Parameter	Default	Optional	Description
cleardigits	"no"	*	Specifies whether previous input should be considered or ignored for the purpose of barge-in and digit matching. When it is set to "yes", any previously buffered digits are discarded. If it is set to "no", previously buffered digits will be considered. If "cleardigits" is set to "no" and "barge" is set to "yes", previously buffered digits will result in the collection phase starting immediately, and the prompt will not be played. Values: • "yes" • "no"
no_cache		*	Cache-control for http URIs. Values: • "yes" does not use caching. • "no" uses caching. The default is set by the configuration.
max_age		*	Cache-control for http URIs, the maximum age, in seconds, of a cached file. The default is set by the configuration.

Parameter	Default	Optional	Description
max_stale		*	Cache-control for http URIs, the number of seconds that a cached file may exceed its expiration time by and still be a considered as fresh. The default is set by the configuration.
fetch_timeout	"300s"	*	http URIs, the maximum time in seconds to retrieve the file. This is the overall period of the transaction.

Request Payload Example


```
<web_service version="1.0">
  <call>
     <call_action>
        <playcollect max_digits="4" timeout="10s" offset="0s" repeat="0" delay="1s"
terminate_digits="#"
                   tone_detection="yes" barge="yes" cleardigits="yes">
                   <play_source audio_uri="file://verification/play_menu.wav"
audio_type="audio/x-wav"/>
        </playcollect>
     </call_action>
  </call>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
            <call_response identifier="c87fcca1-b2d0-49c5-8b89-baaae71cf695"
appid="app"
                           href="http://10.20.129.100:81/default/calls/c87fcca1-b2d0-
49c5-8b89-baaae71cf695"
                           connected="yes" signaling="yes" cpa="no"
call_type="inbound"
                           media="audiovideo"
                           dtmf_mode="rfc2833"
                           destination_uri="sip:sip@10.20.129.100"
                           source_uri="sip:Username@10.20.129.113:5060"
```

```
                           async_dtmf="yes" async_tone="yes" cleardigits="yes"
info_ack_mode="automatic"
                           audio="sendrecv" video="sendrecv">
                           <call_action>
                                <playcollect transaction_id="25608a36-2fd1-43ca-b741-
3420dbc389cb"
                                             fetch_timeout="300s"
                                             terminate_digits="#"
                                             timeout="10s"
                                             interdigit_timeout=""
                                             tone_detection="yes"
                                             max_digits="4"
                                             barge="yes"
                                             offset="0s"
                                             delay="1s"
                                             repeat="0"
                                             cleardigits="yes">
                                             <play_source
audio_uri="file://verification/play_menu.wav"
                                                          audio_type="audio/x-wav"/>
                                </playcollect>
                           </call_action>
          </call_response>
</web_service>
```

overlay

Request Payload Attributes

Parameter	Default	Optional	Description
uri	(none)		The template parameters passed to the image builder. ""image:"id=template&a=b..."
duration	"infinite"	*	The length of time that the overlay is shown. Use "infinite" without the quotes to display the overlay until explicitly stopped.

Request Payload Example

```
<web_service version="1.0">
  <call>
   <call_action>
      <overlay uri="image:id=menu&header=Menu du Jour&items=1 Pie&items=2
Chips&items=3 Burger&items=4
                    Pizza&items=5 Jacket&items=6 Panini&items=7 Pasta&footer=Tuesday"
               duration="15s"/>
   </call_action>
</call></web_service>
```

Response Payload Example

```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
               <overlay transaction_id="c4023bb2-4062-415b-ae79-2e708458cfdd"
                        uri="image:id=menu&header=Menu du Jour&items=1 Pie&items=2
Chips&items=3
                             Burger&items=4 Pizza&items=5 Jacket&items=6
Panini&items=7
                             Pasta&footer=Tuesday"
                        duration="15s"/>
          </call_action>
     </call_response>
</web_service>
```

send_dtmf

Request Payload Attributes

Parameter	Default	Optional	Description
digits			Digit(s) to send [1234567890*#ABCD].
duration	100ms	*	Length of time of each digit.
interval	100ms	*	Time between successive digits.
level	-10dB	*	Amplitude of the dtmf digit tones. Range 0 to -40dB.

Request Payload Example


```
<web_service version="1.0">
  <call>
   <call_action>
      <send_dtmf digits="2345" />
   </call_action>
</call></web_service>
```


Response Payload Example


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
               <send_dtmf transaction_id="c4023bb2-4062-415b-ae79-2e708458cfdd"
                          digits="2345" duration="100ms" interval="100ms" level="-
10dB"/>
          </call_action>
     </call_response>
</web_service>
```

stop

Request Payload Attributes

Parameter	Default	Optional	Description
transaction_id	(none)		Identifier returned by play, playcollect, playrecord, overlay, or send_dtmf call action.

Request Payload Example

```
<web_service version="1.0">
  <call>
    <call_action>
       <stop transaction_id="c4023bb2-4062-415b-ae79-2e708458cfdd"/>
    </call_action>
  </call>
</web_service>
```

Response Payload Example

```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
              <stop transaction_id="c4023bb2-4062-415b-ae79-2e708458cfdd"/>
          </call_action>
     </call_response>
</web_service>
```

add_party/update_party

Request Payload Attributes

Parameter	Default	Optional	Description
conf_id	(none)		Identifier of the conference to join (add_party only).
caption	(none)	*	Text for caption e.g. caller name.
region	"0"	*	The ID of the region used to display this participant's video stream. The value "0" means no preference. The current occupant of the region (if any) will be reset to no preference and replaced by this party.
audio	"recvonly"	*	Sets the conference audio participation. Values: • "inactive" - No audio • "sendonly" - Only transmit audio • "recvonly" - Only receive audio • "sendrecv" - Full duplex audio

Parameter	Default	Optional	Description
video	"recvonly"	*	Sets the conference video participation. Values: • "inactive" - No video • "sendonly" - Only transmit video • "recvonly" - Only receive video • "sendrecv" - Full duplex video
clamp_dtmf	The value specified in create conference	*	Determines if dtmf digits are suppressed. Values: • "yes" • "no"
auto_gain_control	The value specified in create conference	*	Determines if automatic gain control should be used. Values: • "yes" • "no"
echo_cancellation	The value specified create conference	*	Determines if echo cancellation should be used. Values: • "yes" • "no"

Parameter	Default	Optional	Description
mute	"no"	*	Mutes or unmutes the audio stream from this party. Values: • "yes" mute the stream. • "no" unmute the stream.
tx_mute	"no"	*	Mutes or unmutes the audio stream to this party. Values: • "yes" mute the stream. • "no" unmute the stream.
privilege	"no"	*	Enables or disables privilege talker. When set to "yes", party is always included in the conference summation output process, providing its speech level is greater than zero. Values: • "yes" • "no"

Parameter	Default	Optional	Description
mode	"normal"	*	Determines the mixing for the party. Values: "normal" "coach" sets party as a coach, the coach is heard by pupil only. "pupil" sets this party as a pupil, the pupil hears everyone including the coach.

Request Payload Example

add_party

```
<web_service version="1.0">
  <call>
    <call_action>
       <add_party conf_id="830b9fda-d89e-495d-b7a1-6a63402bcdcf" caption="Username"
region="0"
                  audio="sendrecv" video="sendrecv"/>
    </call_action>
  </call>
</web_service>
```

update_party

```
<web_service version="1.0">
  <call>
    <call_action>
       <update_party caption="Frank"/>
    </call_action>
  </call>
</web_service>
```

Response Payload Example

add_party


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
               <add_party conf_id="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
caption="Username" region="0"
                          audio="sendrecv" video="sendrecv"/>
          </call_action>
     </call_response>
</web_service>
```


update_party


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
               <update_party conf_id="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
caption="Frank" region="0"
                             audio="sendrecv" video="sendrecv"/>
          </call_action>
     </call_response>
</web_service>
```

remove_party

Request Payload Example

```
<web_service version="1.0">
  <call>
    <call_action>
       <remove_party />
    </call_action>
  </call>
</web_service>
```

Response Payload Example

```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
              <remove_party conf_id="830b9fda-d89e-495d-b7a1-6a63402bcdcf"/>
          </call_action>
     </call_response>
</web_service>
```

join/unjoin

Request Payload Attributes

Parameter	Default	Optional	Description
call_id	(none)		The identifier of the other party call ID (join only).

Request Payload Example

join

```
<web_service version="1.0">
  <call>
    <call_action>
       <join call_id="830b9fda-d89e-495d-b7a1-6a63402bcdcf"/>
    </call_action>
  </call>
</web_service>
```

unjoin

```
<web_service version="1.0">
  <call>
    <call_action>
       <unjoin/>
    </call_action>
  </call>
</web_service>
```

Response Payload Example

join

```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
               <join call_id="830b9fda-d89e-495d-b7a1-6a63402bcdcf" />
          </call_action>
     </call_response>
</web_service>
```

unjoin

```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
               <unjoin call_id="830b9fda-d89e-495d-b7a1-6a63402bcdcf"/>
          </call_action>
     </call_response>
</web_service>
```

send_info/send_info_ack

Request Payload Attributes

Parameter	Default	Optional	Description
content_type	(none)		Mime type describing content (optional for send_info_ack).
content	(none)		Data content (optional for send_info_ack).

Request Payload Example

send_info

```
<web_service version="1.0">
  <call>
    <call_action>
       <send_info content_type="text/plain" content_data="data"/>
    </call_action>
  </call>
</web_service>
```

send_info_ack

```
<web_service version="1.0">
  <call>
    <call_action>
```

```
      <send_info_ack/>
    </call_action>
  </call>
</web_service>
```


Response Payload Example

send_info


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
               <send_info content_type="text/plain" content_data="data"/>
          </call_action>
     </call_response>
</web_service>
```


send_info_ack


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
               <send_info_ack/>
          </call_action>
     </call_response>
</web_service>
```

transfer

Transfers a call.

- Unattended. The call resource must in a connected state.
- Attended. The call resource must in a connected state and the call identified by call_id must be in a connected state.

Request Payload Attributes

Parameter	Default	Optional	Description
call_id	(none)	*	The call identifier of the consultation call (attended transfer only).
uri	(none)	*	The URI of the transfer target (unattended transfer only).

Request Payload Example

attended

```
<web_service version="1.0">
  <call>
   <call_action>
      <transfer call_id="7de4c7f4-067c-455d-afee-52f57e00314b" />
   </call_action>
</call></web_service>
```

unattended transfer

```
<web_service version="1.0">
  <call>
   <call_action>
      <transfer uri="sip:8001@192.168.195.52" />
   </call_action>
</call></web_service>
```

Response Payload Example

attended transfer

```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
```

```
            source_uri="sip:Username@10.20.129.113:5060"
            async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
            <call_action>
                <transfer call_id="7de4c7f4-067c-455d-afee-52f57e00314b" />
           </call_action>
     </call_response>
</web_service>
```


unattended transfer


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
            href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
            connected="yes" signaling="yes" cpa="no" call_type="inbound"
            media="audio"
            dtmf_mode="rfc2833"
            source_uri="sip:Username@10.20.129.113:5060"
            async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
            <call_action>
                <transfer uri="sip:8001@192.168.195.52" />
           </call_action>
     </call_response>
</web_service>
```


redirect

Redirects an incoming call to another URI.

Request Payload Attributes

Parameter	Default	Optional	Description
uri	(none)		The URI of the redirection target.

Request Payload Example


```
<web_service version="1.0">
  <call>
    <call_action>
       <redirect uri="sip:8001@192.168.195.52"/>
    </call_action>
  </call>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
              <redirect uri="sip:8001@192.168.195.52"/>
          </call_action>
     </call_response>
</web_service>
```


hangup

Ends a call.

- If the call is joined to another call, the other party will be automatically unjoined.
- If the call is in a conference, it will be removed from the conference automatically.

Request Payload Attributes

Parameter	Default	Optional	Description
content_type	(none)	*	Mime type describing content.
content	(none)	*	Data content.

Request Payload Example


```
<web_service version="1.0">
  <call>
    <call_action>
       <hangup content_type="text/plain" content="data"/>
    </call_action>
  </call>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.113:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
              <hangup content_type="text/plain" content="data"/>
          </call_action>
     </call_response>
</web_service>
```


get_call_info

Queries the call information.

Request Payload Example


```
<web_service version="1.0">
  <call>
    <call_action>
       <get_call_info/>
    </call_action>
  </call>
</web_service>
```

Response Payload Attributes

Parameter	Description
local_sdp	Local session description protocol.
remote_sdp	Remote session description protocol.
media	The media types supported by the call. Values: • "audio" • "audiovideo" • "video"
audio	Direction of audio media. Values: • "inactive" • "sendonly" • "recvonly" • "sendrecv"
video	Direction of video media. Values: • "inactive" • "sendonly" • "recvonly" • "sendrecv"
uri	The requested URI.
caller_uri	The callers URI ('To:' header for inbound calls, 'From:' header for outbound calls).
called_uri	The called URI ('From:' header for inbound calls, 'To:' header for outbound calls).
sip_headers	Refer to sip_headers_attributes.
application_id	ID of the controlling application.

Response Payload Example


```
<web_service version="1.0">
     <call_response identifier="8fe4c7f4-067c-455d-afee-52f57e00314b" appid="app"
           href="http://10.20.129.100:81/default/calls/8fe4c7f4-067c-455d-afee-
52f57e00314b"
           connected="yes" signaling="yes" cpa="no" call_type="inbound"
           media="audio"
           dtmf_mode="rfc2833"
           source_uri="sip:Username@10.20.129.115:5060"
           async_dtmf="no" async_tone="no" cleardigits="no" info_ack_mode="automatic"
early_media="yes">
           <call_action>
              <get_call_info local_sdp="[sdp]" remote_sdp="[sdp]" media="audio"
audio="sendrecv" video="sendrecv"
                             uri="sip:sip@10.20.129.97"
called_uri="&ltsip:sip@10.20.129.97&gt"
                             caller_uri="&quotYour Full
Name&quot&ltsip:Username@10.20.129.115:5060l>tag=90DC8D5466B30CBE62A63481AC29B6CD"
                             <sip_headers raw_headers="Call-
ID:FF6844A4360586C46C9468D1CE6AE9031227B@10.20.129.115
                                                       To: &ltsip:sip@10.20.129.97&gt
                                                       From: &quotYour Full
Name&qout&ltsip:Username@10.20.129.115:5060>tag=90DC8D5466B30CBE62A63481AC29B6CD
                                                       Contact:
&ltsip:Username@10.20.129.115:5060;transport=udp&gt
                                                       Content-Type:application/sdp
                                                       Content-Length:454
                                                       User-Agent: Kapanga Softphone
Desktop Windows
                                                                   1.00/2182d+1329238
479_00FFB0EBCB09_A088B4778D9C_2C4138058FBB_CC52AFCC7D8F_08002700A8B4
                                                                   (not registered)
                                                       Session-Expires:
1800;refresher=uac>
                                          <param name="headers.Call-ID"
value="FF6844A4360586C46C9468D1CE6AE9031227B@10.20.129.115"/>
                                          <param name="headers.To"
value="&ltsip:sip@10.20.129.97&gt"/>
                                          <param name="headers.From" value="quotYour
Full Name&qout
                                                                            &ltsip:Us
ername@10.20.129.115:5060>tag=90DC8D5466B30CBE62A63481AC29B6CD"/>
```

```
                                          <param name="headers.Contact"
value="&ltsip:Username@10.20.129.115:5060;transport=udp&gt"/>
                                          <param name="headers.Content-Type"
value="application/sdp"/>
                                          <param name="headers.User-Agent"
value="Kapanga Softphone Desktop Windows
                                                                   1.00/2182d+1329238
479_00FFB0EBCB09_A088B4778D9C_2C4138058FBB_CC52AFCC7D8F_08002700A8B4
                                                                   (not
registered)"/>
                                          <param name="headers.Session-Expires"
value="1800;refresher=uac"/>
                             </sip_header>
              </get_call_info>
          </call_action>
     </call_response>
</web_service>
```


HTTP DELETE

Deletes a call resource.

`DELETE /calls/[call_id]?appid=[app_id]`

Conference Resource

The Conference Resource encapsulates a single instance of a conference resource on PowerMedia XMS. It contains all call resources currently included in the active conference.

Conference-related properties and actions associated with the conference are defined in this section. These include play, update_play, record, and stop.

For details on conference sub-resources, see the Conference Sub-Resource section.

The following tables show the HTTP methods that can be used with a conference.

Note: The payloads shown are examples only as there are many possible variations.

conferences

Resource URI

`/conferences?appid=[app_id]`

HTTP GET

Retrieves all available conference resources.

`GET /conferences?appid=[app_id]`

Response Payload Example


```
<web_service version="1.0">
   <conferences_response size="1">
              <conference_response appid="app" auto_gain_control="yes" beep="yes"
caption="yes"
```

```
                                   caption_duration="30s" clamp_dtmf="yes"
echo_cancellation="yes"
                                   href="http://10.20.129.100:81/default/conferences/
830b9fda-d89e-495d-b7a1-6a63402bcdcf"
                                   identifier="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
layout="2" layout_size="auto"
                                   max_parties="2" reserve="2" type="audiovideo">
              </conference_response>
  </conferences_response>
</web_service>
```

HTTP POST

Creates a conference resource.

`POST /conferences?appid=[app_id]`

Request Payload Attributes

Parameter	Default	Optional	Description
type	"audio"	*	Sets the media supported by conference. Values: • "audio" • "audiovideo"
max_parties	"9"	*	Maximum number of parties in a conference. Range: 2-N
reserve	"0"	*	Number of party resources to reserve for this conference. Any requests beyond this value are honored on a best-effort basis.

Parameter	Default	Optional	Description
layout	"0"	*	The number of regions displayed in the conference video output using a standard layout. Valid values are "0", "1", "2","4", "6" and "9". Setting to "0" means that the number of regions displayed is determined by the number of visible parties. Intermediate values are rounded up to the next supported value e.g. 3 => 4, 7 => 9. Values greater than 9 will result in a 9 region layout.

<table>
<tr><th>Parameter</th><th>Default</th><th>Optional</th><th>Description</th></tr>
<tr><td>layout_regions</td><td></td><td>*</td><td>Specifies a custom video layout. When defined this takes precedence over the layout parameter. The layout is defined a semicolon delimited list of region definitions. Each region is defined using the format: <ID> EQUALS <LEFT> COMMA <TOP> COMMA <RELATIVE_SIZE>[COMMA PRIORITY].
ID uniquely identifies the region. Note, the value "0" is reserved and may not be used as a region identifier.
The LEFT and TOP parameters define the position of the top left corner of the region as an offset from the top left corner of the root window, they are expressed as percentages.
RELATIVE_SIZE defines the size of a region relative to the size of the root window either a fraction or decimal percentage.
The PRIORITY parameter is optional and is an integer values from 0 to 10, where 0 means disabled and 1 is the highest priority (default).
Example: "left = 5, 5, 50, 2; right = 45, 45, 1/2, 1".</td></tr>
</table>

Parameter	Default	Optional	Description
layout_size	"auto"	*	Determines the size of the root window. Values: • "auto" • "qcif" • "cif" • "vga" • "720p" The value "auto" causes the window size to track the size of the largest party.
caption	"yes"	*	Determines if the caller's ID is overlaid on their image. Values: • "yes" • "no"
caption_duration	"20s"	*	The length of time that the caption is shown. Use "infinite" without the quotes to display the caption for the entire call.
beep	"yes"	*	Determines if a tone is played when a party joins/leaves a conference. Values: • "yes" • "no"
clamp_dtmf	"yes"	*	Determines if DTMF digits are suppressed. Values: • "yes" • "no"

Parameter	Default	Optional	Description
auto_gain_control	"yes"	*	Determines if automatic gain control should be used. Values: • "yes" • "no"
echo_cancellation	"yes"	*	Determines if echo cancellation should be used. Values: • "yes" • "no"
active_talker_region	(none)	*	Specifies the ID of the region used to display the active talker's video.
active_talker_interval	"0"	*	Specifies the minimum duration of time that must pass before changes to active talkers will be notified. The minimum value is 500ms. A value of "0" disables active talker notifications.
max_active_talkers	"10"	*	Sets the maximum number of active talkers to be included in the audio mix. Values range from 2 to 10 inclusive.

Request Payload Example

```
<web_service version="1.0">
  <conference type="audiovideo" max_parties="2" reserve="2" layout="2"
              caption="yes" caption_duration="30s" beep="yes" clamp_dtmf="yes"
              auto_gain_control="yes" echo_cancellation="yes"/>
</web_service>
```

Response Payload Example

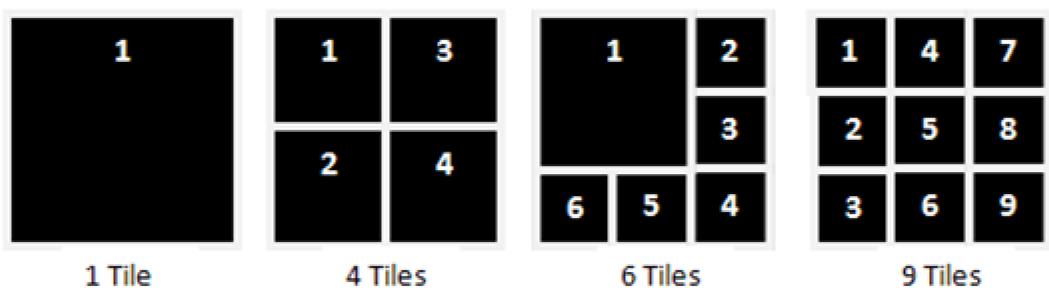

```
<web_service version="1.0">
              <conference_response appid="app" auto_gain_control="yes" beep="yes"
caption="yes" caption_duration="30s"
                                   clamp_dtmf="yes" echo_cancellation="yes"
                                   href="http://10.20.129.100:81/default/conferences/
830b9fda-d89e-495d-b7a1-6a63402bcdcf"
                                   identifier="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
layout="2" layout_size="auto"
                                   max_parties="2" reserve="2" type="audiovideo">
              </conference_response>
</web_service>
```


Conference Concepts

This section contains a higher-level look at various aspects of PowerMedia XMS conference behavior.

Screen Layout

The standard video conference tiling layout for 1, 4, 6 and 9 tile conferences and the order in which conferees appear in the tiles is shown below:



Conference Sub-Resource

For details on conference resources, see the Conference Resource section.

conference

Resource URI

`/conferences/[conference_id]?appid=[app_id]`

HTTP GET

Retrieves an available conference resource.

`GET /conferences/[conference_id]?appid=[app_id]`

Response Payload Example


```
<web_service version="1.0">
              <conference_response appid="app" auto_gain_control="yes" beep="yes"
caption="yes" caption_duration="30s"
                                   clamp_dtmf="yes" echo_cancellation="yes"
                                   href="http://10.20.129.100:81/default/conferences/
830b9fda-d89e-495d-b7a1-6a63402bcdcf"
                                   identifier="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
layout="2" layout_size="auto"
                                   max_parties="2" reserve="2" type="audiovideo">
                                   <conf_participant audio="sendrecv"
auto_gain_control="yes"
                                                     call_id="c87fcca1-b2d0-49c5-
8b89-baaae71cf695"
                                                     caption="Username"
clamp_dtmf="yes" echo_cancellation="yes"
                                                     mode="normal" mute="no"
privilege="no" region="0"
                                                     tx_mute="no" video="sendrecv"/>
              </conference_response>
</web_service>
```


HTTP PUT

Updates a conference resource.

```
PUT /conferences/[conference_id]?appid=[app_id]
```

Modify Conference Attributes

Parameter	Default	Optional	Description
layout	(none)	*	The number of tiles displayed in the conference output. Valid values are "0", "1", "2", "4", "6" and "9". Setting to "0" means that the number of tiles displayed is determined by the number of active presenters.

Parameter	Default	Optional	Description
layout_regions	(none)	*	Specifies a custom video layout. When defined this takes precedence over the layout parameter. The layout is defined a semicolon delimited list of region definitions. Each region is defined using the format: <ID> EQUALS <LEFT> COMMA <TOP> COMMA <RELATIVE_SIZE>[COMMA PRIORITY]. • ID uniquely identifies the region. Note, the value "0" is reserved and may not be used as a region identifier. • The LEFT and TOP parameters define the position of the top left corner of the region as an offset from the top left corner of the root window, they are expressed as percentages. • RELATIVE_SIZE defines the size of a region relative to the size of the root window either a fraction or decimal percentage. • The PRIORITY parameter is optional and is an integer values from 0 to 10, where 0 means disabled and 1 is the highest priority (default). Example: "left = 5, 5, 50, 2; right = 45, 45, 1/2, 1"

Parameter	Default	Optional	Description
layout_size	(none)	*	Determines the size of the root window. Values: • "auto" • "qcif" • "cif" • "vga" • "720p" The value "auto" causes the window size to track the size of the largest party.
active_talker_region	(none)	*	Specifies the ID of the region used to display the active talker's video. Use "0" to disable.
active_talker_interval	(none)	*	Specifies the minimum duration of time that must pass before changes to active talkers will be notified. The minimum value is 500ms. A value of "0" disables active talker notifications.
max_active_talkers	(none)	*	Sets the maximum number of active talkers to be included in the audio mix. Values range from 2 to 10 inclusive. Note, it is an error to set the value lower than the current number of privileged parties.

Request Payload Example

```
<web_service version="1.0">
  <conference layout_size="vga"/>
</web_service>
```

Response Payload Example


```
<web_service version="1.0">
              <conference_response appid="app" auto_gain_control="yes" beep="yes"
caption="yes" caption_duration="30s"
                                   clamp_dtmf="yes" echo_cancellation="yes"
                                   href="http://10.20.129.100:81/default/conferences/
830b9fda-d89e-495d-b7a1-6a63402bcdcf"
                                   identifier="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
layout="2" layout_size="vga"
                                   max_parties="2" reserve="2" type="audiovideo">
                                   <conf_participant audio="sendrecv"
auto_gain_control="yes"
                                                     call_id="c87fcca1-b2d0-49c5-
8b89-baaae71cf695"
                                                     caption="Username"
clamp_dtmf="yes" echo_cancellation="yes"
                                                     mode="normal" mute="no"
privilege="no" region="0"
                                                     tx_mute="no" video="sendrecv"/>
              </conference_response>
</web_service>
```


Perform Conference Action

- play
- update_play
- record
- stop

play

Request Payload Attributes

Parameter	Default	Optional	Description
play_source	(none)		Refer to play_source_attributes.
offset	"0"	*	Specifies the time offset from where the play should start. Note, the 'offset' is applied to the initial play only.

Parameter	Default	Optional	Description
repeat	"0"	*	Number of times to repeat the play. Use "infinite" to repeat indefinitely. "file://" URIs only.
delay	"1s"	*	Time delay between repeated plays.
max_time	"infinite"	*	Limit the playback time to this value.
region	"0"	*	The ID of the region that will display the video media. The value "0" causes the video to be shown full-screen with the current layout being restored automatically when the play back completes.
no_cache		*	Cache-control for http URIs. Values: • "yes" does not use caching. • "no" uses caching. The default is set by the configuration.
max_age		*	Cache-control for http URIs, the maximum age, in seconds, of a cached file. The default is set by the configuration.

Parameter	Default	Optional	Description
max_stale		*	Cache-control for http URIs, the number of seconds that a cached file may exceed its expiration time by and still be a considered as fresh. The default is set by the configuration.
fetch_timeout	"300s"	*	http URIs, the maximum time in seconds to retrieve the file. This is the overall period of the transaction.

Request Payload Example

```
<web_service version="1.0">
  <conference>
    <conf_action>
        <play max_time="20s" >
              <play_source audio_uri="file://verification/play_menu.wav"
audio_type="audio/x-wav" />
        </play>
    </conf_action>
</conference>
</web_service>
```

Response Payload Example

```
<web_service version="1.0">
        <conference_response identifier="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
appid="app"
                             href="http://10.20.129.100:81/default/conferences/830b9f
da-d89e-495d-b7a1-6a63402bcdcf"
                             type="audiovideo" max_parties="2" reserve="2"
                             layout="2" caption="yes" caption_duration="30s"
beep="yes"
                             clamp_dtmf="yes" auto_gain_control="yes"
echo_cancellation="yes" layout_size="auto">
                             <conf_participant call_id="c87fcca1-b2d0-49c5-8b89-
baaae71cf695" audio="sendrecv"
                                               video="sendrecv" caption="Username"
region="0"      clamp_dtmf="yes"
```

```
                                               auto_gain_control="yes"
echo_cancellation="yes"  mute="no" tx_mute="no"
                                               privilege="no" mode="normal"/>
                             <conf_action>
                                 <play transaction_id="acb08f84-afb4-430b-92f2-
22083b7638aa"
                                       max_time="20s"
                                       fetch_timeout="300s"
                                       offset="0s"
                                       delay="1s"
                                       repeat="0"
                                       region="0">
                                       <play_source
audio_uri="file://verification/play_menu.wav"
                                                     audio_type="audio/x-wav"/>
                                 </play>
                             </conf_action>
         </conference_response>
</web_service>
```


update_play

Request Payload Attributes

Parameter	Default	Optional	Description
transaction_id	(none)		Media identifier, returned by play.
dvr_action	(none)		Values: • "backward" - skip backwards. • "forward" - skip forward. • "pause" - pause playback. • "restart" - jump back to the start. • "resume" - resume paused playback.

Request Payload Example


```
<web_service version="1.0">
  <conference>
    <conf_action>
         <update_play transaction_id="acb08f84-afb4-430b-92f2-22083b7638aa"
dvr_action="pause"/>
    </conf_action>
  </conference>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
        <conference_response identifier="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
appid="app"
                             href="http://10.20.129.100:81/default/conferences/830b9f
da-d89e-495d-b7a1-6a63402bcdcf"
                             type="audiovideo" max_parties="2" reserve="2"
                             layout="2" caption="yes" caption_duration="30s"
beep="yes"
                             clamp_dtmf="yes" auto_gain_control="yes"
echo_cancellation="yes" layout_size="auto">
                             <conf_participant call_id="c87fcca1-b2d0-49c5-8b89-
baaae71cf695" audio="sendrecv"
                                              video="sendrecv" caption="Username"
region="0"      clamp_dtmf="yes"
                                              auto_gain_control="yes"
echo_cancellation="yes"  mute="no" tx_mute="no"
                                              privilege="no" mode="normal"/>
                             <conf_action>
                                 <update_play transaction_id="acb08f84-afb4-430b-
92f2-22083b7638aa"
                                             dvr_action="pause"/>
                             </conf_action>
         </conference_response>
</web_service>
```

record

Request Payload Attributes

Parameter	Default	Optional	Description
recording_audio_mime_params	(none)		Refer to recording_audio_mime_params.
recording_video_mime_params	(none)		Refer to recording_video_mime_params.
terminate_digits	"#"	*	The digit or digits [0-9,*,#] used to terminate the play_record.
recording_uri	(none)		A filename "file://..." which must refer to an existing directory or "http://" ("https://"). See also media DEPRECATED.
recording_audio_uri	(none)		The URL of the audio media to be recorded e.g. "file://...", "http(s)://..."
recording_audio_type	(none)		The mime-type of the audio media. • "audio/x-wav" • "audio/basic" • "audio/x-alaw-basic"
recording_video_uri	(none)		The URL of the video media to be recorded e.g. "file://...", "http(s)://..."
recording_video_type	(none)		The mime-type of the video media. • "video/x-vid"
max_time	"infinite"	*	The maximum length of time to record. Use "infinite" for no limit.

Request Payload Example


```
<web_service version="1.0">
  <conference>
      <conf_action>
              <record recording_audio_uri="file://recorded_file.wav"
recording_audio_type="audio/x-wav"
                            max_time="10s"  terminate_digits="#"  />
      </conf_action>
  </conference>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
        <conference_response identifier="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
appid="app"
                             href="http://10.20.129.100:81/default/conferences/830b9f
da-d89e-495d-b7a1-6a63402bcdcf"
                             type="audiovideo" max_parties="2" reserve="2"
                             layout="2" caption="yes" caption_duration="30s"
beep="yes"
                             clamp_dtmf="yes" auto_gain_control="yes"
echo_cancellation="yes" layout_size="auto">
                             <conf_participant call_id="c87fcca1-b2d0-49c5-8b89-
baaae71cf695" audio="sendrecv"
                                              video="sendrecv" caption="Username"
region="0"      clamp_dtmf="yes"
                                              auto_gain_control="yes"
echo_cancellation="yes"  mute="no" tx_mute="no"
                                              privilege="no" mode="normal"/>
                             <conf_action>
                                 <record transaction_id="a560cbc1-5674-44e8-bb21-
2b34130169c4"
                                         terminate_digits="#"
                                         max_time="10s"

recording_audio_uri="file://recorded_file.wav"
                                         recording_audio_type="audio/x-
wav"/>
                             </conf_action>
         </conference_response>
</web_service>
```

stop

Request Payload Attributes

Parameter	Default	Optional	Description
transcation_id	(none)		Identifier returned by play or record conference action.

Request Payload Example

```
<web_service version="1.0">
  <conference>
    <conf_action>
      <stop transaction_id="c4023bb2-4062-415b-ae79-2e708458cfdd"/>
    </conf_action>
  </conference>
</web_service>
```

Response Payload Example

```
<web_service version="1.0">
        <conference_response identifier="830b9fda-d89e-495d-b7a1-6a63402bcdcf"
appid="app"
                             href="http://10.20.129.100:81/default/conferences/830b9f
da-d89e-495d-b7a1-6a63402bcdcf"
                             type="audiovideo" max_parties="2" reserve="2"
                             layout="2" caption="yes" caption_duration="30s"
beep="yes"
                             clamp_dtmf="yes" auto_gain_control="yes"
echo_cancellation="yes" layout_size="auto">
                             <conf_participant call_id="c87fcca1-b2d0-49c5-8b89-
baaae71cf695" audio="sendrecv"
                                               video="sendrecv" caption="Username"
region="0"      clamp_dtmf="yes"
                                               auto_gain_control="yes"
echo_cancellation="yes"  mute="no" tx_mute="no"
                                               privilege="no" mode="normal"/>
                             <conf_action>
                                  <stop transaction_id="c4023bb2-4062-415b-ae79-
2e708458cfdd"/>
                             </conf_action>
         </conference_response>
</web_service>
```

HTTP DELETE

Deletes a conference resource.

```
DELETE /conferences/[conference_id]?appid=[app_id]
```

Event Handler Resource

HTTP event streaming is implemented in the PowerMedia XMS RESTful server as an eventhandler resource. When the client wishes to receive asynchronous events, it uses the web service to create an eventhandler and to subscribe to specific event types.

For example, when the client performs an HTTP GET on a newly created eventhandler, the PowerMedia XMS RESTful server responds with a `200 OK`; however, the TCP connection remains open until the client destroys the eventhandler. Event data related to resources and subscribed event types are sent to the client until it deletes the eventhandler. Event data related to resources and subscribed event types are sent to the client until it deletes the eventhandler.

An enabled PowerMedia XMS application ID must be included in the URL for the original HTTP POST where the eventhandler is created. This is utilized to assure that clients only have access to resources they create.

For details on eventhandler sub-resources, see the Event Handler Sub-Resource section.

The following tables show the HTTP methods that can be used with the eventhandler.

Note: The payloads shown are examples only as there are many possible variations.

RESTful Event Streaming Data Format Change

As of PowerMedia XMS Release 2.2 Service Update 5, the RESTful event format has been updated to be compliant with HTTP chunked data formatting (RFC 7230, Section 4.1). The extra carriage return / line feed (CRLF) in previous PowerMedia XMS versions has been removed from the beginning of each chunk. Each chunk returned begins with the size of the XML payload in hex format.

Example:

```
44
<web_service version="1.0">
<event type="keepalive"/></web_service>
```

Note: Existing RESTful applications that make use of event handlers will require updating.

eventhandlers

Resource URI

```
/eventhandlers?appid=[app_id]
```

eventsubscribe_attributes

Parameter	Default	Optional	Description
action	add	*	This will add/remove an event subscription. Values: • "add" • "remove"
type	any	*	Type of events to monitor. Values: • "end_play" • "end_record" • "end_playcollect" • "end_playrecord" • "end_overlay" • "end_dtmf" • "keepalive" • "incoming" • "ringing" • "connected" • "hangup" • "info" • "dtmf" • "tone" • "any" • "end_speak" • "start_of_input" • "end_recognize" • "answered" • "active_talker" (any: any event type)
resource_id	any	*	Monitor events for a specific resource. (any: any resource id)

Parameter	Default	Optional	Description
resource_type	any	*	Monitor events for a specific resource type. Values: • "call" • "conference" • "mrcp" • "any" (any: any resource type)

HTTP GET

Retrieves all available eventhandler resources.

`GET /eventhandlers?appid=[app_id]`

Response Payload Example


```
<web_service version="1.0">
        <eventhandlers_response size="1">
                <eventhandler_response appid="app"
                                          href="http://10.20.129.100:81/default/eventhandl
ers/2d013b6d-c943-46d9-a7d5-
                                                8509cfb177ce"
                                          identifier="2d013b6d-c943-46d9-a7d5-
8509cfb177ce">
                                          <eventsubscribe resource_id="any"
resource_type="any" type="any"/>
                </eventhandler_response>
        </eventhandlers_response>
</web_service>
```


HTTP POST

Creates an eventhandler resource.

`POST /eventhandlers?appid=[app_id]`

Request Payload Attributes

Parameter	Default	Optional	Description
eventsubscribe		*	Refer to eventsubscribe_attributes.

Request Payload Example


```
<web_service version="1.0">
    <eventhandler>
        <eventsubscribe  type="any" resource_id="any" resource_type="any"/>
    </eventhandler>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
      <eventhandler_response identifier="90ba2f00-e460-4351-9203-1a2fcf25a739"
appid="app"
                             href="http://10.20.129.100:81/default/eventhandlers/90ba
2f00-e460-4351-9203-1a2fcf25a739">
                             <eventsubscribe type="any" resource_id="any"
resource_type="any"/>
      </eventhandler_response>
</web_service>
```


Event Handler Sub-Resource

For details on eventhandler resources, see the Event Handler Resource section.

eventhandler

Resource URI

`/eventhandlers/[eventhandler_id]?appid=[app_id]`

HTTP GET

Get the events.

`GET /eventhandlers/[eventhandler_id]?appid=[app_id]`

event_data_attributes

Parameter	Default	Optional	Description
name	(none)		Event data name.
value	(none)		Event data value. Refer to the Events section for details on what is provided in each event.

Response Payload Attributes

Parameter	Default	Optional	Description
type	(none)		Type of events to monitor. Values: • "end_play" • "end_record" • "end_playcollect" • "end_playrecord" • "end_overlay" • "end_dtmf" • "keepalive" • "incoming" • "ringing" • "connected" • "hangup" • "info" • "dtmf" • "tone" • "any" • "end_speak" • "start_of_input" • "end_recognize" • "answered" • "active_talker"
resource_id	(none)	*	Monitor events for a specific resource.
resource_type	(none)	*	Monitor events for a specific resource type. Values: • "call" • "conference" • "mrcp" • "any"
event_data	(none)	*	Values (if applicable) for all data that describe each event. Refer to event_data_attributes or the Events section for details on what is provided in each event. This can be repeated 0 ~ n.

Response Payload Example

T1:

```
200 OK
```

T2:


```
256
<web_service version="1.0">
    <event type="hangup" resource_id="c87fcca1-b2d0-49c5-8b89-baaae71cf695"
resource_type="call">
           <event_data name="reason" value="5800 IPEC_SIPReasonStatusBYE" />
    </event>
</web_service>
```


T3:


```
345
<web_service version="1.0">
    <event type="incoming" resource_id="603bf73e-5e74-4c72-865a-6e498a5e2ad5"
resource_type="call">
           <event_data name="caller_uri" value="sip:Username@10.20.129.113:5060" />
           <event_data name="uri" value="sip:sip@10.20.129.100" />
    </event>
</web_service>
```


T(N):

```
0
```

- in T2 time: the first line is the size of payload, in above example size is 256.
- in T(N) time: size is 0 it means no more events and the connection about to close.

HTTP PUT

Adds or removes an event subscription.

```
PUT /eventhandlers/[eventhandler_id]?appid=[app_id]
```

Request Payload Attributes

Parameter	Default	Optional	Description
eventsubscribe	(none)		Refer to eventsubscribe_attributes.

Request Payload Example


```
<web_service version="1.0">
  <eventhandler>
        <eventsubscribe action="add" type="incoming" resource_id="any"
resource_type="any"/>
  </eventhandler>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
   <eventhandler_response identifier="2d013b6d-c943-46d9-a7d5-8509cfb177ce"
appid="app"
                          href="http://10.20.129.100:81/default/eventhandlers/2d013b6
d-c943-46d9-a7d5-8509cfb177ce">
                          <eventsubscribe type="any" resource_id="any"
resource_type="any"/>
                          <eventsubscribe type="incoming" resource_id="any"
resource_type="any"/>
   </eventhandler_response>
</web_service>
```


HTTP DELETE

Deletes an eventhandler resource.

```
DELETE /eventhandlers/[eventhandler_id]?appid=[app_id]
```

Events

This section describes the event data that is associated with event types. Events are asynchronously returned to the application from the eventhandler.

This section describes the events that are provided by PowerMedia XMS to a RESTful application on the open TCP connection maintained between PowerMedia XMS and the RESTful application. Events are asynchronously sent to the application via the eventhandler.

keepalive

Once the application starts to monitor the events, the REST Web Service will send a "keepalive" event periodically.

Event Payload Example


```
<web_service version="1.0">
             <event type="keepalive" />
</web_service>
```

Media Events

end_play

Completion event for play (conference_play and call_play).

- **transaction_id**
- **reason**
  - "end"
  - "stopped"
  - "max-time"
  - "error"
  - "hangup"
- **duration** - in milliseconds.
- **status** - extended error information.

Event Payload Example


```
<web_service version="1.0">
     <event type="end_play" resourceid="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5"
resource_type="call">
            <event_data name="reason" value="complete" />
            <event_data name="duration" vaule="30000ms" />
            <event_data name="transaction_id" value="0974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efdd" />
     </event>
</web_service>
```


end_playcollect

Completion event for playcollect (call_playcollect).

- **transaction_id**
- **reason**
  - "max-digits"
  - "term-digit"
  - "timeout"
  - "tone"
  - "stopped"
  - "hangup"
- **digits** - digits collected.
- **tone** - tone identifier (if reason is "tone").
- **duration** - in milliseconds.
- **status** - extended error information.

Event Payload Example


```
<web_service version="1.0">
      <event type="end_playcollect" resourceid="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5"
resource_type="call">
             <event_data name="reason" value="term-digit" />
             <event_data name="digits" vaule="1234" />
             <event_data name="transaction_id" value="0974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efdd" />
      </event>
</web_service>
```


end_record

Completion event for record (call_record).

- **transaction_id**
- **reason**
  - "term-digit"
  - "timeout"
  - "stopped"
  - "max-time"
  - "max-silence"
  - "hangup"
- **duration** - in milliseconds.
- **audio_location** - from a response to HTTP PUT, this is the location header value.
- **video_location** - from a response to HTTP PUT, this is the location header value.
- **status** - extended error information.

Event Payload Example


```
<web_service version="1.0">
        <event type="end_record" resource_id="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5"
resource_type="call">
               <event_data name="reason" value="timeout" />
               <event_data name="transaction_id" value="0974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efdd" />
        </event>
</web_service>
```

end_playrecord

Completion event for playrecord (call_playrecord).

- **transaction_id**
- **reason**
  - "term-digit"
  - "timeout"
  - "stopped"
  - "max-time"
  - "max-silence"
  - "hangup"
- **duration** - in milliseconds.
- **audio_location** - from a response to HTTP PUT, this is the location header value.
- **video_location** - from a response to HTTP PUT, this is the location header value.
- **status** - extended error information.

Event Payload Example


```
<web_service version="1.0">
        <event type="end_playrecord" resource_id="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5" resource_type="call">
               <event_data name="reason" value="timeout" />
               <event_data name="transaction_id" value="0974c8b5-8a3c-4a8e-ae82-8f7c8bd0efdd" />
        </event>
</web_service>
```


end_overlay

Completion event for overlay (call_overlay).

- **transaction_id**
- **reason**
  - "stopped"
  - "max-time"
- **duration** - in milliseconds.

Event Payload Example


```
<web_service version="1.0">
            <event type="end_overlay" resource_id="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5" resource_type="call">
                   <event_data name="reason" value=" stop " />
                   <event_data name="transaction_id" value="0974c8b5-8a3c-4a8e-ae82-8f7c8bd0efdd" />
            </event>
</web_service>
```

Call Events

incoming

A new inbound call.

- **call_id**
- **uri**
- **caller_uri**
- **name** - application name, from routing rule.
- **headers** - raw SIP headers, delimited by the <CR><LF> end-of-line characters.
- **headers.<NAME>** - individual SIP header.
- **content_type** - mime type of content.
- **content** - optional content.

Event Payload Example

```
<web_service version="1.0">
            <event type="incoming" resource_id="5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" resource_type="call">
                   <event_data name="call_id" value=" 5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" />
                   <event_data name="uri" value="sip:sip@10.20.129.8" />
                   <event_data name="caller_uri" value="sip:frank@10.20.129.20" />
            </event>
</web_service>
```

ringing

The remote party of an outbound call is ringing.

- **call_id**

Event Payload Example

```
<web_service version="1.0">
            <event type="ringing" resource_id="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5"
resource_type="call">
                   <event_data name="call_id" value=" 5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" />
            </event>
</web_service>
```

connected

The remote party of an outbound call has answered.

- **call_id**
- **reason**
    - "unknown"
    - "answer-machine"
    - "voice"
    - "ced" (fax detection)
    - custom tone name
- **media**
    - "audio"
    - "audiovideo"
- **caller_uri** ('From' header)
- **called_uri** ('To' header)

Event Payload Example


```
<web_service version="1.0">
            <event type="connected" resource_id="5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" resource_type="call">
                  <event_data name="reason" value="voice" />
                  <event_data name="media" value="audio" />
            </event>
</web_service>
```


hangup

An outbound call request has failed or the remote party has ended the call.

All call resources are released automatically, and if the call is joined to another call, the other party will be automatically unjoined. No further actions may be performed on the call.

- **call_id**
- **reason**
    - "busy-tone"
    - "operator-intercept"
    - "no-answer"
- **content_type**
- **content**

Event Payload Example


```
<web_service version="1.0">
            <event type="hangup" resource_id="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5"
resource_type="call">
                  <event_data name="reason" value="no-answer" />
            </event>
</web_service>
```

info

Unsolicited user information (for example, SIP INFO).

- **call_id**
- **content_type**
- **content**

Event Payload Example


```
<web_service version="1.0">
             <event type="info" resource_id="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5"
resource_type="call">
                    <event_data name="call_id" value=" 5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" />
                    <event_data name="content-type" value="sdp" />
                    <event_data name="content" value="[sdp]" />
             </event>
</web_service>
```


dtmf

Unsolicited DTMF digits.

- **call_id**
- **digits**

Event Payload Example


```
<web_service version="1.0">
             <event type="dtmf" resource_id="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5"
resource_type="call">
                    <event_data name="call_id" value=" 5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" />
                    <event_data name="digits" value="4" />
             </event>
</web_service>
```


tone

Unsolicited tone detection events.

- **call_id**
- **tone**

Event Payload Example


```
<web_service version="1.0">
             <event type="tone" resource_id="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5"
resource_type="call">
                    <event_data name="call_id" value="5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" />
                    <event_data name="tone" value="[name]" />
             </event>
</web_service>
```


updated

Indicates that the call's media has been updated (for example, SIP reINVITE).

- **call_id**
- **media**
  - "unknown"
  - "audio"
  - "video"
  - "audiovideo"
- **audio**
  - "inactive"
  - "sendonly"
  - "recvonly"
  - "sendrecv"
- **video**
  - "inactive"
  - "sendonly"
  - "recvonly"
  - "sendrecv"

Event Payload Example


```
<web_service version="1.0">
             <event type="tone" resource_id="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5"
resource_type="call">
                    <event_data name="call_id" value="5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" />
                    <event_data name="media" value="audiovideo" />
                    <event_data name="audio" value="sendrecv" />
                    <event_data name="video" value="sendrecv" />
             </event>
</web_service>
```

MRCP Events

end_speak

Completion event for mrcp_speak.

- **mrcp_id**
- **id**
- **transaction_id**
- **reason** - raw result from MRCP server.
- **duration** - in milliseconds.

Event Payload Example

```
<web_service version="1.0">
             <event type="tone" resource_id="97d105ae-30b9-45e2-acdf-82e67d9b34aa"
resource_type="mrcp">
                    <event_data name="id" value="5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" />
                    <event_data name="mrcp_id" value="97d105ae-30b9-45e2-acdf-
82e67d9b34aa" />
                    <event_data name="transaction_id" value="0974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efdd" />
                    <event_data name="reason" value="anything" />
                    <event_data name="duration" value="20s" />
             </event>
</web_service>
```

end_recognize

Completion event for mrcp_recognize.

- **mrcp_id**
- **id**
- **transaction_id**
- **reason** - raw result from MRCP server.
- **duration** - in milliseconds.
- **content**
- **content_type**
- **waveform_uri**

Event Payload Example

```
<web_service version="1.0">
             <event type="tone" resource_id="97d105ae-30b9-45e2-acdf-82e67d9b34aa"
resource_type="mrcp">
                    <event_data name="id" value="5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" />
                    <event_data name="mrcp_id" value="97d105ae-30b9-45e2-acdf-
82e67d9b34aa" />
                    <event_data name="transaction_id" value="0974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efdd" />
                    <event_data name="reason" value="anything" />
                    <event_data name="duration" value="20s" />
                    <event_data name="content" value="data" />
                    <event_data name="content_type" value="plain/text" />
             </event>
</web_service>
```

start_of_input

Input detected event for mrcp_recognize.

- **mrcp_id**
- **id**
- **transaction_id**

Event Payload Example

```
<web_service version="1.0">
             <event type="tone" resource_id="97d105ae-30b9-45e2-acdf-82e67d9b34aa"
resource_type="mrcp">
                    <event_data name="id" value="5974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efd5" />
                    <event_data name="mrcp_id" value="97d105ae-30b9-45e2-acdf-
82e67d9b34aa" />
                    <event_data name="transaction_id" value="0974c8b5-8a3c-4a8e-ae82-
8f7c8bd0efdd" />
             </event>
</web_service>
```

Conference Events

active_talker

- **conf_id**
- **talkers** - space delimited list of call IDs.

Event Payload Example


```
<web_service version="1.0">
             <event type="active_talker" resource_id="97d105ae-30b9-45e2-acdf-82e67d9b34aa" resource_type="conference">
                    <event_data name="conf_id" value="97d105ae-30b9-45e2-acdf-82e67d9b34aa" />
                    <event_data name="talkers" value="5974c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5 6798c8b5-8a3c-4a8e-ae82-8f7c8bd0efd5" />
             </event>
</web_service>
```


MRCP Resource

The Media Resource Control Protocol (MRCP), is used by PowerMedia XMS as an interface to Automatic Speech Recognition (ASR) and Text-to-Speech (TTS) systems. MRCP provides an easy way to build voice user interfaces, allowing a grammar to be built for speech input and providing a way to easily translate text into voice prompts without reading and recording them.

For details on MRCP sub-resources, see the MRCP Sub-Resource section.

The following tables show the HTTP methods that can be used with MRCP.

Note: The payloads shown are examples only as there are many possible variations.

mrcps

Resource URI

`/mrcps?appid=[app_id]`

HTTP GET

Retrieves all available MRCP resources.

`GET /mrcps?appid=[app_id]`

Response Payload Example


```
<web_service version="1.0">
   <mrcps_response size="1">
              <mrcps_response size="1">
                     <mrcp_response appid="app" asr="yes" tts="yes"
                                    href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-9e0b-6d9f05fb4ee3"
                                    identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3">
                     </mrcp_response>
              </mrcps_response>
  </mrcps_response>
</web_service>
```

HTTP POST

Creates a MRCP resource.

`POST /mrcps?appid=[app_id]`

Request Payload Attributes

Parameter	Default	Optional	Description
asr	"yes"	*	Specifies if a speech recognizer resource is required. Values: • "yes" • "no"
tts	"yes"	*	Specifies if a speech synthesizer resource is required. Values: • "yes" • "no"

Request Payload Example

```
<web_service version="1.0">
  <mrcp asr="yes" tts="yes"/>
</web_service>
```

Response Payload Example

```
<web_service version="1.0">
              <mrcp_response appid="app" asr="yes" tts="yes"
                             href="http://10.20.129.100:81/default/mrcps/8d200e91-
8b56-465d-9e0b-6d9f05fb4ee3"
                             identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3">
              </mrcp_response>
</web_service>
```

MRCP Sub-Resource

For details on MRCP resources, see the MRCP Resource section.

mrcp

Resource URI

`/mrcps/[mrcp_id]?appid=[app_id]`

HTTP GET

Retrieves an available MRCP resource.

`GET /mrcps/[mrcp_id]?appid=[app_id]`

Response Payload Example


```
<web_service version="1.0">
            <mrcp_response appid="app" asr="yes" tts="yes"
                           href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-9e0b-6d9f05fb4ee3"
                          identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3">
            </mrcp_response>
</web_service>
```


HTTP PUT

Updates a MRCP resource.

`PUT /mrcps/[mrcp_id]?appid=[app_id]`

Perform MRCP Action

- speak
- recognize
- set-asr-param/set-tts-param
- get-asr-param/get-tts-param
- define-grammar
- mrcp-update-action

param_attributes

Parameter	Default	Optional	Description
name	(none)		Parameter name.
value	(none)		Parameter value.

speak

Synthesizes speech (TTS).

Request Payload Attributes

Parameter	Default	Optional	Description
call_id	(none)		The call receiving the synthesized speech.
param	(none)	*	Generic parameter. Refer to param_attributes can be repeated 0 ~ n.
barge	"yes"	*	Sets whether the synthesis can be barged. Values: • "yes" • "no"
locale	"en-US"	*	Language and country code. See RFC 3066.
content	(none)		Content to be synthesized.
content_type	(none)		Mime type of the content.

Request Payload Example

```
<web_service version="1.0">
  <mrcp>
     <mrcp_action>
          <speak call_id="c87fcca1-b2d0-49c5-8b89-baaae71cf695" barge="yes"
content="Hello World" content_type="text/plain"/>
     </mrcp_action>
  </mrcp>
</web_service>
```

Response Payload Example


```
<web_service version="1.0">
       <mrcp_response identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3" appid="app"
                      href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-
9e0b-6d9f05fb4ee3"
                      asr="yes" tts="yes">
                      <mrcp_action>
                         <speak transaction_id="b9e5c859-e5ed-44fd-bf91-b636ba8b14f0"
                                call_id="c87fcca1-b2d0-49c5-8b89-baaae71cf695"
                                barge="yes"
                                locale="en-US"
                                content="Hello World"
                                content_type="text/plain">
                         </speak>
                      </mrcp_action>
       </mrcp_response>
</web_service>
```


recognize

Recognizes speech (ASR).

Request Payload Attributes

Parameter	Default	Optional	Description
call_id	(none)		The call sourcing the media to be recognized.
param	(none)	*	Generic parameter. Refer to param_attributes can be repeated 0 ~ n.
grammar	(none)	*	Grammar for recognizer.
grammar_type	(none)	*	Mime type of the grammar.
grammar_id	(none)	*	Application identifier for a previously defined grammar.
timeout	"infinite"	*	The maximum length of time to wait for input.

Request Payload Example

```
<web_service version="1.0">
 <mrcp>
    <mrcp_action>
         <recognize call_id="c87fcca1-b2d0-49c5-8b89-baaae71cf695"
                    grammar="grammar" grammar_type="type" grammar_id="text/plain"
timeout="infinite"/>
    </mrcp_action>
 </mrcp>
</web_service>
```

Response Payload Example

```
<web_service version="1.0">
       <mrcp_response identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3" appid="app"
                      href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-
9e0b-6d9f05fb4ee3"
                      asr="yes" tts="yes">
                      <mrcp_action>
                         <recognize transaction_id="842722ef-75d0-407d-9b52-
29c5270868bb"
                                    call_id="c87fcca1-b2d0-49c5-8b89-baaae71cf695"
                                    grammar="grammar"
                                    grammar_type="type"
                                    grammar_id="text/plain"
                                    timeout="infinite">
                         </recognize>
                      </mrcp_action>
       </mrcp_response>
</web_service>
```

set-asr-param/set-tts-param

Request Payload Attributes

Parameter	Default	Optional	Description
param	(none)	*	Generic parameter. Value has the format: name=value, name2=value2

Request Payload Example

set-asr-param

```
<web_service version="1.0">
  <mrcp>
      <mrcp_action>
          <set-asr-param>
               <param name="name" value="value"/>
          </set-asr-param>
      </mrcp_action>
  </mrcp>
</web_service>
```

set-tts-param

```
<web_service version="1.0">
  <mrcp>
      <mrcp_action>
          <set-tts-param>
               <param name="name" value="value"/>
          </set-tts-param>
      </mrcp_action>
  </mrcp>
</web_service>
```

Response Payload Example

set-asr-param

```
<web_service version="1.0">
       <mrcp_response identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3" appid="app"
                      href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-
9e0b-6d9f05fb4ee3"
                      asr="yes" tts="yes">
                      <mrcp_action>
                         <set-asr-param>
                             <param name="name" value="value"/>
                         </set-asr-param>
                      </mrcp_action>
       </mrcp_response>
</web_service>
```

set-tts-param

```
<web_service version="1.0">
       <mrcp_response identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3" appid="app"
                      href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-9e0b-6d9f05fb4ee3"
                      asr="yes" tts="yes">
                      <mrcp_action>
                         <set-tts-param>
                             <param name="name" value="value"/>
                         </set-tts-param>
                      </mrcp_action>
       </mrcp_response>
</web_service>
```

get-asr-param/get-tts-param

Request Payload Example

get-asr-param

```
<web_service version="1.0">
  <mrcp>
      <mrcp_action>
          <get-asr-param/>
      </mrcp_action>
  </mrcp>
</web_service>
```

get-tts-param

```
<web_service version="1.0">
  <mrcp>
      <mrcp_action>
          <get-tts-param/>
      </mrcp_action>
  </mrcp>
</web_service>
```

Response Payload Example

get-asr-param


```
<web_service version="1.0">
       <mrcp_response identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3" appid="app"
                      href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-9e0b-6d9f05fb4ee3"
                      asr="yes" tts="yes">
                      <mrcp_action>
                         <get-asr-param>
                             <param name="name" value="value"/>
                         </get-asr-param>
                      </mrcp_action>
       </mrcp_response>
</web_service>
```


get-tts-param


```
<web_service version="1.0">
       <mrcp_response identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3" appid="app"
                      href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-9e0b-6d9f05fb4ee3"
                      asr="yes" tts="yes">
                      <mrcp_action>
                         <get-tts-param>
                             <param name="name" value="value"/>
                         </get-tts-param>
                      </mrcp_action>
       </mrcp_response>
</web_service>
```


define-grammar

Request Payload Attributes

Parameter	Default	Optional	Description
grammar	(none)	*	Grammar for recognizer.
grammar_type	(none)	*	Mime type of the grammar.
grammar_id	(none)	*	Application identifier for a previously defined grammar.

Request Payload Example

```
<web_service version="1.0">
 <mrcp>
    <mrcp_action>
         <define-grammar  grammar="grammar" grammar_type="type" grammar_id="id"/>
    </mrcp_action>
 </mrcp>
</web_service>
```

Response Payload Example

```
<web_service version="1.0">
       <mrcp_response identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3" appid="app"
                      href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-
9e0b-6d9f05fb4ee3"
                      asr="yes" tts="yes">
                      <mrcp_action>
                         <define-grammar  grammar="grammar" grammar_type="type"
grammar_id="id"/>
                      </mrcp_action>
       </mrcp_response>
</web_service>
```

mrcp-update-action

Request Payload Attributes

Parameter	Default	Optional	Description
action	(none)		Values: * stop-speak * stop-recognize * pause (speak only) * resume (speak only) * barge (speak only) * start-input-timers (recognize only)

Request Payload Example


```
<web_service version="1.0">
 <mrcp>
    <mrcp_action>
         <mrcp-update-action action="pause"/>
    </mrcp_action>
 </mrcp>
</web_service>
```


Response Payload Example


```
<web_service version="1.0">
       <mrcp_response identifier="8d200e91-8b56-465d-9e0b-6d9f05fb4ee3" appid="app"
                      href="http://10.20.129.100:81/default/mrcps/8d200e91-8b56-465d-
9e0b-6d9f05fb4ee3"
                      asr="yes" tts="yes">
                      <mrcp_action>
                         <mrcp-update-action action="pause"/>
                      </mrcp_action>
       </mrcp_response>
</web_service>
```


HTTP DELETE

Deletes a MRCP resource.

```
DELETE /mrcps/[mrcp_id]?appid=[app_id]
```

6. XML Schema Definition of Elements

This section contains the complete XML schema definition of elements (XSD).

Note: This schema definition may occasionally be updated. Always use the XSD (*xmsrest.xsd*) available with the current PowerMedia XMS version in the */etc/xms* directory.

```
<?xml version="1.0" encoding="UTF-8" ?>
<xs:schema xmlns:xs="http://www.w3.org/2001/XMLSchema" elementFormDefault="qualified">

	<xs:simpleType name="boolean_type">
		<xs:restriction base="xs:string">
			<xs:enumeration value="yes" />
			<xs:enumeration value="no" />
		</xs:restriction>
	</xs:simpleType>

	<xs:simpleType name="action_option">
		<xs:restriction base="xs:string">
			<xs:enumeration value="add" />
			<xs:enumeration value="remove" />
		</xs:restriction>
	</xs:simpleType>

	<xs:simpleType name="call_type_option">
		<xs:restriction base="xs:string">
			<xs:enumeration value="inbound" />
			<xs:enumeration value="outbound" />
			<xs:enumeration value="3pcc" />
		</xs:restriction>
	</xs:simpleType>

	<xs:simpleType name="media_type">
		<xs:restriction base="xs:string">
			<xs:enumeration value="audio" />
			<xs:enumeration value="video" />
			<xs:enumeration value="audiovideo" />
			<xs:enumeration value="unknown" />
		</xs:restriction>
	</xs:simpleType>

	<xs:simpleType name="media_direction">
		<xs:restriction base="xs:string">
			<xs:enumeration value="inactive" />
			<xs:enumeration value="sendonly" />
			<xs:enumeration value="recvonly" />
			<xs:enumeration value="sendrecv" />
		</xs:restriction>
	</xs:simpleType>

	<xs:simpleType name="audio_codec_option">
		<xs:restriction base="xs:string">
			<xs:enumeration value="L16" />
			<xs:enumeration value="mulaw" />
			<xs:enumeration value="alaw" />
		</xs:restriction>
	</xs:simpleType>

	<xs:simpleType name="audio_rate_option">
		<xs:restriction base="xs:string">
			<xs:enumeration value="8000" />
			<xs:enumeration value="16000" />
		</xs:restriction>
	</xs:simpleType>

	<xs:simpleType name="video_codec_option">
		<xs:restriction base="xs:string">
```

```
                    <xs:enumeration value="h264" />
                    <xs:enumeration value="h263" />
                    <xs:enumeration value="mp4v-es" />
            </xs:restriction>
    </xs:simpleType>

    <xs:simpleType name="video_type_option">
            <xs:restriction base="xs:string">
                    <xs:enumeration value="video/x-vid" />
                    <xs:enumeration value="image/jpeg" />
            </xs:restriction>
    </xs:simpleType>

    <xs:simpleType name="audio_type_option">
            <xs:restriction base="xs:string">
                    <xs:enumeration value="audio/x-wav" />
                    <xs:enumeration value="audio/basic" />
                    <xs:enumeration value="audio/x-alaw-basic" />
                    <xs:enumeration value="audio/G723" />
                    <xs:enumeration value="audio/G726" />
                    <xs:enumeration value="audio/G729" />
            </xs:restriction>
    </xs:simpleType>

    <xs:simpleType name="recording_video_type_option">
            <xs:restriction base="xs:string">
                    <xs:enumeration value="video/x-vid" />
            </xs:restriction>
    </xs:simpleType>



    <xs:simpleType name="conf_party_mode">
            <xs:restriction base="xs:string">
                    <xs:enumeration value="normal" />
                    <xs:enumeration value="coach" />
                    <xs:enumeration value="pupil" />
            </xs:restriction>
    </xs:simpleType>

    <xs:simpleType name="layout_size_option">
            <xs:restriction base="xs:string">
                    <xs:enumeration value="automatic" />
                    <xs:enumeration value="qcif" />
                    <xs:enumeration value="cif" />
                    <xs:enumeration value="vga" />
                    <xs:enumeration value="720p" />
            </xs:restriction>
    </xs:simpleType>

    <xs:simpleType name="rtp_encryption_option">
            <xs:restriction base="xs:string">
                    <xs:enumeration value="none" />
                    <xs:enumeration value="dtls" />
            </xs:restriction>
    </xs:simpleType>

    <xs:simpleType name="time_value">
            <xs:restriction base="xs:string">
                    <xs:pattern value="(\+)?([0-9]*\.)?[0-9]+(ms|s)|infinite"/>
            </xs:restriction>
</xs:simpleType>

<xs:simpleType name="digit_value">
            <xs:restriction base="xs:string">
                    <xs:pattern value="[0-9#*]+|"/>
            </xs:restriction>
    </xs:simpleType>

    <xs:simpleType name="rfc2833_digit_value">
            <xs:restriction base="xs:string">
```

```
                        <xs:pattern value="[0-9#*a-dA-D]+|"/>
                </xs:restriction>
        </xs:simpleType>

        <xs:simpleType name="dtmf_tone_range">
                <xs:restriction base="xs:string">
                        <xs:pattern value="(0|(\-)([0-9]|1[0-9]|2[0-9]|3[0-9]|40))(dB|db|DB|Db)"/>
                </xs:restriction>
        </xs:simpleType>

        <xs:simpleType name="volume_range">
                <xs:restriction base="xs:string">
                        <xs:pattern value="((\+|)([0-9]|1[0-9]|2[0-9]|3[0-1])|(\-)([0-9]|1[0-
9]|2[0-9]|3[0-2]))(dB|db|DB|Db)"/>
                </xs:restriction>
        </xs:simpleType>

        <xs:simpleType name="integer_value">
                <xs:restriction base="xs:string">
                        <xs:pattern value="[0-9]+|infinite"/>
                </xs:restriction>
        </xs:simpleType>

        <xs:simpleType name="dtmf_mode_option">
                <xs:restriction base="xs:string">
                        <xs:enumeration value="inband" />
                        <xs:enumeration value="outofband" />
                        <xs:enumeration value="rfc2833" />
                </xs:restriction>
        </xs:simpleType>

        <xs:simpleType name="info_ack_mode_option">
                <xs:restriction base="xs:string">
                        <xs:enumeration value="automatic" />
                        <xs:enumeration value="manual" />
                </xs:restriction>
        </xs:simpleType>

        <xs:simpleType name="event_type">
                <xs:restriction base="xs:string">
                        <xs:enumeration value="end_play" />
                        <xs:enumeration value="end_record" />
                        <xs:enumeration value="end_playcollect" />
                        <xs:enumeration value="end_playrecord" />
                        <xs:enumeration value="end_overlay" />
                        <xs:enumeration value="end_dtmf" />
                        <xs:enumeration value="keepalive" />
                        <xs:enumeration value="incoming" />
                        <xs:enumeration value="ringing" />
                        <xs:enumeration value="connected" />
                        <xs:enumeration value="hangup" />
             <xs:enumeration value="info" />
                        <xs:enumeration value="dtmf" />
                        <xs:enumeration value="tone" />
                        <xs:enumeration value="any" />
                        <xs:enumeration value="end_speak" />
                        <xs:enumeration value="start_of_input" />
                        <xs:enumeration value="end_recognize" />
                        <xs:enumeration value="answered" />
                        <xs:enumeration value="updated" />
                        <xs:enumeration value="active_talker" />
                </xs:restriction>
        </xs:simpleType>

        <xs:simpleType name="event_data_name">
                <xs:restriction base="xs:string">
                        <xs:enumeration value="tone" />
                        <xs:enumeration value="digits" />
                        <xs:enumeration value="info" />
                        <xs:enumeration value="reason" />
                        <xs:enumeration value="duration" />
```

```
                <xs:enumeration value="uri" />
                <xs:enumeration value="caller_uri" />
                <xs:enumeration value="content_type" />
                <xs:enumeration value="content" />
                <xs:enumeration value="transaction_id" />
            </xs:restriction>
        </xs:simpleType>

        <xs:simpleType name="event_resource_type">
            <xs:restriction base="xs:string">
                <xs:enumeration value="call" />
                <xs:enumeration value="conference" />
                <xs:enumeration value="mrcp" />
                <xs:enumeration value="any" />
            </xs:restriction>
        </xs:simpleType>

        <xs:element name="dvr_setting">
            <xs:complexType>
                <xs:attribute name="forward_key" type="digit_value" default="1" />
                <xs:attribute name="backward_key" type="digit_value" default="2" />
                <xs:attribute name="pause_key" type="digit_value" default="3" />
                <xs:attribute name="resume_key" type="digit_value" default="4" />
                <xs:attribute name="restart_key" type="digit_value" default="5" />
            </xs:complexType>
        </xs:element>

        <xs:simpleType name="dvr_action_option">
            <xs:restriction base="xs:string">
                <xs:enumeration value="forward" />
                <xs:enumeration value="backward" />
                <xs:enumeration value="pause" />
                <xs:enumeration value="resume" />
                <xs:enumeration value="restart" />
            </xs:restriction>
        </xs:simpleType>

        <xs:simpleType name="mrcp_action_option">
            <xs:restriction base="xs:string">
                <xs:enumeration value="stop-speak" />
                <xs:enumeration value="stop-recognize" />
                <xs:enumeration value="pause" />
                <xs:enumeration value="resume" />
                <xs:enumeration value="barge" />
                <xs:enumeration value="start-input-timers" />
            </xs:restriction>
        </xs:simpleType>
        <xs:element name="recording_audio_mime_params">
            <xs:complexType>
                    <xs:attribute name="codec" type="audio_codec_option" />
                    <xs:attribute name="rate" type="audio_rate_option" />
            </xs:complexType>
        </xs:element>

        <xs:element name="recording_video_mime_params">
            <xs:complexType>
                    <xs:attribute name="codec" type="video_codec_option" />
                    <xs:attribute name="profile" type="digit_value" />
                    <xs:attribute name="level" type="xs:string" />
                    <xs:attribute name="framerate" type="digit_value" />
                    <xs:attribute name="maxbitrate" type="digit_value" />
                    <xs:attribute name="height" type="digit_value" />
                    <xs:attribute name="width" type="digit_value" />
            </xs:complexType>
        </xs:element>

        <xs:element name="param">
            <xs:complexType>
                    <xs:attribute name="name" type="xs:string" use="required"/>
                    <xs:attribute name="value" type="xs:string" use="required"/>
            </xs:complexType>
```

```
        </xs:element>

        <xs:element name="conf_participant">
                <xs:complexType>
                                <xs:attribute name="call_id" type="xs:string" />
                                <xs:attribute name="audio" type="media_direction" />
                                <xs:attribute name="video" type="media_direction" />
                                <xs:attribute name="caption" type="xs:string" />
                                <xs:attribute name="region" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="add_party">
                <xs:complexType>
                                <xs:attribute name="conf_id" type="xs:string" use="required" />
                                <xs:attribute name="audio" type="media_direction"
default="recvonly" />
                                <xs:attribute name="video" type="media_direction"
default="recvonly" />
                                <xs:attribute name="caption" type="xs:string" />
                                <xs:attribute name="clamp_dtmf" type="boolean_type" default="yes"
/>
                                <xs:attribute name="auto_gain_control" type="boolean_type"
default="yes" />
                                <xs:attribute name="echo_cancellation" type="boolean_type"
default="yes" />
                                <xs:attribute name="mute" type="boolean_type" default="no" />
                                <xs:attribute name="tx_mute" type="boolean_type" default="no" />
                                <xs:attribute name="privilege" type="boolean_type" default="no" />
                                <xs:attribute name="mode" type="conf_party_mode" default="normal"
/>
                                <xs:attribute name="region" type="xs:string" default="0"/>
                </xs:complexType>
        </xs:element>

        <xs:element name="update_party">
                <xs:complexType>
                                <xs:attribute name="conf_id" type="xs:string" />
                                <xs:attribute name="audio" type="media_direction" />
                                <xs:attribute name="video" type="media_direction" />
                                <xs:attribute name="caption" type="xs:string" />
                                <xs:attribute name="clamp_dtmf" type="boolean_type" />
                                <xs:attribute name="auto_gain_control" type="boolean_type"/>
                                <xs:attribute name="echo_cancellation" type="boolean_type" />
                                <xs:attribute name="mute" type="boolean_type"  />
                                <xs:attribute name="tx_mute" type="boolean_type"  />
                                <xs:attribute name="privilege" type="boolean_type"  />
                                <xs:attribute name="mode" type="conf_party_mode"  />
                                <xs:attribute name="region" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="remove_party">
                <xs:complexType>
                                <xs:attribute name="conf_id" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="join">
                <xs:complexType>
                                <xs:attribute name="call_id" type="xs:string" use="required" />
                </xs:complexType>
        </xs:element>

        <xs:element name="unjoin">
                <xs:complexType>
                                <xs:attribute name="call_id" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="transfer">
```

```
                <xs:complexType>
                                <xs:attribute name="call_id" type="xs:string" />
                                <xs:attribute name="uri" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="redirect">
                <xs:complexType>
                                <xs:attribute name="uri" type="xs:string" use="required"/>
                </xs:complexType>
        </xs:element>

        <xs:element name="hangup">
                <xs:complexType>
                                <xs:attribute name="content_type" type="xs:string" />
                                <xs:attribute name="content" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="send_info">
                <xs:complexType>
                                <xs:attribute name="content_type" type="xs:string" use="required"/>
                                <xs:attribute name="content" type="xs:string" use="required"/>
                </xs:complexType>
        </xs:element>

        <xs:element name="send_info_ack">
                <xs:complexType>
                                <xs:attribute name="content_type" type="xs:string" />
                                <xs:attribute name="content" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="send_dtmf">
                <xs:complexType>
                                <xs:attribute name="digits" type="rfc2833_digit_value"
use="required"/>
                                <xs:attribute name="duration" type="time_value" default="100ms"/>
                                <xs:attribute name="interval" type="time_value" default="100ms"/>
                                <xs:attribute name="level" type="dtmf_tone_range" default="-10dB"/>
                                <xs:attribute name="transaction_id" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="get_call_info">
                <xs:complexType>
                                <xs:all minOccurs="0">
                                        <xs:element ref="sip_headers" />
                                </xs:all>
                                <xs:attribute name="local_sdp" type="xs:string" />
                                <xs:attribute name="remote_sdp" type="xs:string" />
                                <xs:attribute name="media" type="media_type"/>
                                <xs:attribute name="audio" type="media_direction" />
                                <xs:attribute name="video" type="media_direction" />
                                <xs:attribute name="uri" type="xs:string" />
                                <xs:attribute name="caller_uri" type="xs:string" />
                                <xs:attribute name="called_uri" type="xs:string" />
                                <xs:attribute name="application_id" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="call_action">
                <xs:complexType>
                        <xs:choice minOccurs="1" maxOccurs="1">
                                <xs:element ref="play" />
                                <xs:element ref="record" />
                                <xs:element ref="update_play" />
                                <xs:element ref="playcollect" />
                                <xs:element ref="playrecord" />
                                <xs:element ref="overlay" />
                                <xs:element ref="stop" />
```

```
                        <xs:element ref="join" />
                        <xs:element ref="unjoin" />
                        <xs:element ref="add_party" />
                        <xs:element ref="update_party" />
                        <xs:element ref="remove_party" />
                        <xs:element ref="send_dtmf" />
                        <xs:element ref="send_info" />
                        <xs:element ref="send_info_ack" />
                        <xs:element ref="transfer" />
                        <xs:element ref="redirect" />
                        <xs:element ref="hangup" />
                        <xs:element ref="get_call_info" />
                </xs:choice>
        </xs:complexType>
</xs:element>

<xs:element name="conf_action">
        <xs:complexType>
                <xs:choice minOccurs="1" maxOccurs="1">
                        <xs:element ref="play" />
                        <xs:element ref="record" />
                        <xs:element ref="update_play" />
                        <xs:element ref="stop" />
                </xs:choice>
        </xs:complexType>
</xs:element>

<xs:element name="speak">
        <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="param" minOccurs="0" maxOccurs="unbounded"
/>
                        </xs:sequence>
                        <xs:attribute name="call_id" type="xs:string" use="required" />
                        <xs:attribute name="barge" type="boolean_type" default="yes" />
                        <xs:attribute name="locale" type="xs:string" default="en-US" />
                        <xs:attribute name="content" type="xs:string" use="required" />
                        <xs:attribute name="content_type" type="xs:string" use="required"
/>
                        <xs:attribute name="transaction_id" type="xs:string" />
        </xs:complexType>
</xs:element>

<xs:element name="recognize">
        <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="param" minOccurs="0" maxOccurs="unbounded"
/>
                        </xs:sequence>
                        <xs:attribute name="call_id" type="xs:string" use="required" />
                        <xs:attribute name="grammar" type="xs:string" />
                        <xs:attribute name="grammar_type" type="xs:string" />
                        <xs:attribute name="grammar_id" type="xs:string" />
                        <xs:attribute name="timeout" type="time_value" default="infinite"
/>
                        <xs:attribute name="transaction_id" type="xs:string" />
        </xs:complexType>
</xs:element>

<xs:element name="set-asr-param">
        <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="param" minOccurs="1" maxOccurs="unbounded"
/>
                        </xs:sequence>
        </xs:complexType>
</xs:element>

<xs:element name="get-asr-param">
        <xs:complexType>
                        <xs:sequence>
```

```
                                        <xs:element ref="param" minOccurs="0" maxOccurs="unbounded"
/>
                                </xs:sequence>
                        </xs:complexType>
                </xs:element>

                <xs:element name="set-tts-param">
                        <xs:complexType>
                                <xs:sequence>
                                        <xs:element ref="param" minOccurs="1" maxOccurs="unbounded"
/>
                                </xs:sequence>
                        </xs:complexType>
                </xs:element>

                <xs:element name="define-grammar">
                        <xs:complexType>
                                <xs:attribute name="grammar" type="xs:string" use="required"/>
                                <xs:attribute name="grammar_type" type="xs:string" use="required"/>
                                <xs:attribute name="grammar_id" type="xs:string" use="required"/>
                        </xs:complexType>
                </xs:element>

                <xs:element name="get-tts-param">
                        <xs:complexType>
                                <xs:sequence>
                                        <xs:element ref="param" minOccurs="0" maxOccurs="unbounded"
/>
                                </xs:sequence>
                        </xs:complexType>
                </xs:element>

                <xs:element name="mrcp-update-action">
                        <xs:complexType>
                                <xs:attribute name="action" type="mrcp_action_option"
use="required"/>
                        </xs:complexType>
                </xs:element>

                <xs:element name="mrcp_action">
                        <xs:complexType>
                                <xs:choice minOccurs="1" maxOccurs="1">
                                        <xs:element ref="speak" />
                                        <xs:element ref="recognize" />
                                        <xs:element ref="mrcp-update-action" />
                                        <xs:element ref="set-asr-param" />
                                        <xs:element ref="get-asr-param" />
                                        <xs:element ref="set-tts-param" />
                                        <xs:element ref="get-tts-param" />
                                        <xs:element ref="define-grammar" />
                                </xs:choice>
                        </xs:complexType>
                </xs:element>

                <xs:attributeGroup name="response_attrgroup">
                        <xs:attribute name="href" type="xs:string" use="required" />
                        <xs:attribute name="identifier" type="xs:string" use="required" />
                        <xs:attribute name="appid" type="xs:string" use="required" />
                </xs:attributeGroup>

                <xs:element name="overlay">
                        <xs:complexType>
                                <xs:attribute name="uri" type="xs:string" use="required"/>
                                <xs:attribute name="duration" type="time_value" default="infinite" />
                                <xs:attribute name="transaction_id" type="xs:string" />
                        </xs:complexType>
                </xs:element>

                <xs:element name="event_data">
                        <xs:complexType>
                                <xs:attribute name="name" type="xs:string" use="required" />
```

```
                <xs:attribute name="value" type="xs:string" use="required" />
        </xs:complexType>
</xs:element>


<xs:element name="sip_headers">
        <xs:complexType>
                <xs:sequence>
                        <xs:element ref="param" minOccurs="0" maxOccurs="unbounded" />
                </xs:sequence>
                <xs:attribute name="raw_sip_headers" type="xs:string"  />
        </xs:complexType>
</xs:element>

<xs:element name="event">
        <xs:complexType>
                <xs:sequence>
                        <xs:element ref="event_data" minOccurs="0" maxOccurs="unbounded" />
                </xs:sequence>
                <xs:attribute name="type" type="event_type" use="required" />
                <xs:attribute name="resource_type" type="xs:string" />
                <xs:attribute name="resource_id" type="xs:string" />
        </xs:complexType>
</xs:element>
<xs:element name="stop">
        <xs:complexType>
                <xs:attribute name="transaction_id" type="xs:string" use="required"/>
        </xs:complexType>
</xs:element>

<xs:element name="play_source">
        <xs:complexType>
                <xs:attribute name="location" type="xs:string" />
                <xs:attribute name="base_audio_uri" type="xs:string" />
                <xs:attribute name="audio_uri" type="xs:string" />
                <xs:attribute name="audio_type" type="audio_type_option" />
                <xs:attribute name="base_video_uri" type="xs:string" />
                <xs:attribute name="video_uri" type="xs:string" />
                <xs:attribute name="video_type" type="video_type_option" />
        </xs:complexType>
</xs:element>

<xs:element name="play">
        <xs:complexType>
                <xs:sequence>
                        <xs:element ref="play_source" minOccurs="1" maxOccurs="1" />
                        <xs:element ref="dvr_setting" minOccurs="0" maxOccurs="1" />
                </xs:sequence>
                <xs:attribute name="offset" type="time_value" default="0s" />
                <xs:attribute name="repeat" type="integer_value" default="0" />
                <xs:attribute name="delay" type="time_value" default="1s" />
                <xs:attribute name="skip_interval" type="time_value" default="1s" />
                <xs:attribute name="max_time" type="time_value" default="infinite" />
                <xs:attribute name="terminate_digits" type="digit_value" default="#"/>
                <xs:attribute name="region" type="xs:string" />
                <xs:attribute name="transaction_id" type="xs:string" />
                <xs:attribute name="no_cache" type="boolean_type" />
                <xs:attribute name="max_age" type="time_value" />
                <xs:attribute name="max_stale" type="time_value" />
                <xs:attribute name="fetch_timeout" type="time_value" default="300s" />
        </xs:complexType>
</xs:element>

<xs:element name="update_play">
        <xs:complexType>
                <xs:attribute name="dvr_action" type="dvr_action_option" use="required"/>
                <xs:attribute name="transaction_id" type="xs:string" use="required"/>
        </xs:complexType>
</xs:element>

<xs:element name="record">
```

```
                <xs:complexType>
                        <xs:all>
                                <xs:element ref="recording_audio_mime_params" minOccurs="0"
maxOccurs="1" />
                                <xs:element ref="recording_video_mime_params" minOccurs="0"
maxOccurs="1" />
                        </xs:all>
                        <xs:attribute name="terminate_digits" type="digit_value" default="#"/>
            <xs:attribute name="recording_uri" type="xs:string" />
            <xs:attribute name="recording_audio_uri" type="xs:string" />
                        <xs:attribute name="recording_audio_type" type="audio_type_option" />
                        <xs:attribute name="recording_video_uri" type="xs:string" />
                        <xs:attribute name="recording_video_type"
type="recording_video_type_option" default="video/x-vid"/>
                        <xs:attribute name="max_silence" type="time_value" default="infinite"  />
                        <xs:attribute name="max_time" type="time_value" default="infinite" />
                        <xs:attribute name="noinput_timeout" type="time_value" default="infinite"
/>
                        <xs:attribute name="transaction_id" type="xs:string" />
                </xs:complexType>
        </xs:element>

        <xs:element name="playrecord">
                <xs:complexType>
            <xs:all>
                                <xs:element ref="play_source" minOccurs="0" maxOccurs="1" />
                                <xs:element ref="recording_audio_mime_params" minOccurs="0"
maxOccurs="1" />
                                <xs:element ref="recording_video_mime_params" minOccurs="0"
maxOccurs="1" />
                        </xs:all>
                        <xs:attribute name="barge" type="boolean_type" default="yes" />
                        <xs:attribute name="cleardigits" type="boolean_type" default="no" />
                        <xs:attribute name="offset" type="time_value" default="0s" />
                        <xs:attribute name="repeat" type="integer_value" default="0" />
                        <xs:attribute name="delay" type="time_value" default="1s" />
                        <xs:attribute name="recording_uri" type="xs:string" />
                        <xs:attribute name="recording_audio_uri" type="xs:string" />
                        <xs:attribute name="recording_audio_type" type="audio_type_option" />
                        <xs:attribute name="recording_video_uri" type="xs:string" />
                        <xs:attribute name="recording_video_type"
type="recording_video_type_option" default="video/x-vid"/>
                        <xs:attribute name="beep" type="boolean_type" default="yes" />
                        <xs:attribute name="terminate_digits" type="digit_value" default="#"/>
                        <xs:attribute name="max_time" type="time_value" default="infinite" />
                        <xs:attribute name="max_silence" type="time_value" default="infinite"  />
                        <xs:attribute name="noinput_timeout" type="time_value"
default="infinite"  />
                        <xs:attribute name="transaction_id" type="xs:string" />
                        <xs:attribute name="no_cache" type="boolean_type" />
                        <xs:attribute name="max_age" type="time_value" />
                        <xs:attribute name="max_stale" type="time_value" />
                        <xs:attribute name="fetch_timeout" type="time_value" default="300s" />
                </xs:complexType>
        </xs:element>

        <xs:element name="playcollect">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="play_source" minOccurs="0" maxOccurs="1" />
                        </xs:sequence>
                        <xs:attribute name="barge" type="boolean_type" default="yes" />
                        <xs:attribute name="cleardigits" type="boolean_type" default="no" />
                        <xs:attribute name="offset" type="time_value" default="0s" />
                        <xs:attribute name="repeat" type="integer_value" default="0" />
                        <xs:attribute name="delay" type="time_value" default="1s" />
                        <xs:attribute name="max_digits" type="xs:string" />
                        <xs:attribute name="timeout" type="time_value" />
                        <xs:attribute name="interdigit_timeout" type="time_value" />
                        <xs:attribute name="terminate_digits" type="digit_value" default="#"/>
                        <xs:attribute name="tone_detection" type="boolean_type" default="no" />
```

```
                <xs:attribute name="transaction_id" type="xs:string" />
                <xs:attribute name="no_cache" type="boolean_type" />
                <xs:attribute name="max_age" type="time_value" />
                <xs:attribute name="max_stale" type="time_value" />
                <xs:attribute name="fetch_timeout" type="time_value" default="300s" />
        </xs:complexType>
</xs:element>

<xs:element name="error">
        <xs:complexType>
                <xs:attribute name="code" type="xs:string" use="required" />
                <xs:attribute name="description" type="xs:string" use="required" />
        </xs:complexType>
</xs:element>

<xs:element name="call">
        <xs:complexType>
                <xs:sequence>
                        <xs:element ref="call_action" minOccurs="0" maxOccurs="1" />
                </xs:sequence>
     <xs:attribute name="answer" type="boolean_type"/>
                <xs:attribute name="signaling" type="boolean_type" default="yes" />
                <xs:attribute name="media" type="media_type" default= "audio"/>
                <xs:attribute name="source_uri" type="xs:string" />
                <xs:attribute name="destination_uri" type="xs:string" />
                <xs:attribute name="called_uri" type="xs:string" />
                <xs:attribute name="display_name" type="xs:string"/>
                <xs:attribute name="sdp" type="xs:string"/>
                <xs:attribute name="cpa" type="boolean_type" default="no"/>
                <xs:attribute name="dtmf_mode" type="dtmf_mode_option" default="rfc2833"/>
                <xs:attribute name="async_dtmf" type="boolean_type" />
                <xs:attribute name="async_tone" type="boolean_type" />
                <xs:attribute name="rx_delta" type="volume_range" />
                <xs:attribute name="tx_delta" type="volume_range" />
                <xs:attribute name="cleardigits" type="boolean_type" />
                <xs:attribute name="info_ack_mode" type="info_ack_mode_option" />
                <xs:attribute name="early_media" type="boolean_type" />
                <xs:attribute name="accept" type="boolean_type" />
                <xs:attribute name="async_completion" type="boolean_type" />
                <xs:attribute name="dial_timeout" type="time_value"/>
                <xs:attribute name="encryption" type="rtp_encryption_option" />
                <xs:attribute name="ice" type="boolean_type"/>
                <xs:attribute name="content" type="xs:string" />
                <xs:attribute name="content_type" type="xs:string" />
        </xs:complexType>
</xs:element>
<xs:element name="call_response">
        <xs:complexType>
                <xs:sequence>
                        <xs:element ref="call_action" minOccurs="0" />
                </xs:sequence>
                <xs:attribute name="signaling" type="boolean_type" />
                <xs:attribute name="media" type="media_type" />
     <xs:attribute name="destination_uri" type="xs:string" />
     <xs:attribute name="display_name" type="xs:string"/>
     <xs:attribute name="source_uri" type="xs:string" />
     <xs:attribute name="called_uri" type="xs:string" />
                <xs:attribute name="call_type" type="call_type_option" />
                <xs:attribute name="connected" type="boolean_type" />
     <xs:attribute name="sdp" type="xs:string"/>
                <xs:attribute name="cpa" type="boolean_type" />
                <xs:attribute name="dtmf_mode" type="dtmf_mode_option" />
                <xs:attribute name="async_dtmf" type="boolean_type"/>
                <xs:attribute name="async_tone" type="boolean_type" />
                <xs:attribute name="rx_delta" type="volume_range" />
                <xs:attribute name="tx_delta" type="volume_range" />
                <xs:attribute name="cleardigits" type="boolean_type" />
                <xs:attribute name="info_ack_mode" type="info_ack_mode_option" />
                <xs:attribute name="early_media" type="boolean_type" />
                <xs:attribute name="audio" type="media_direction" />
                <xs:attribute name="video" type="media_direction" />
```

```
                        <xs:attribute name="async_completion" type="boolean_type" />
                        <xs:attribute name="encryption" type="rtp_encryption_option" />
                        <xs:attribute name="ice" type="boolean_type"/>
                        <xs:attribute name="content" type="xs:string" />
                        <xs:attribute name="content_type" type="xs:string" />
                        <xs:attributeGroup ref="response_attrgroup" />
                </xs:complexType>
        </xs:element>
        <xs:element name="calls_response">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="call_response" minOccurs="0"
maxOccurs="unbounded"/>
                        </xs:sequence>
                        <xs:attribute name="size" type="xs:string" use="required" />
                </xs:complexType>
        </xs:element>

        <xs:element name="eventhandler">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="eventsubscribe" minOccurs="1"
maxOccurs="unbounded" />
                        </xs:sequence>
                </xs:complexType>
        </xs:element>

        <xs:element name="eventhandler_response">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="eventsubscribe" minOccurs="1"
maxOccurs="unbounded" />
                        </xs:sequence>
                        <xs:attributeGroup ref="response_attrgroup" />
                </xs:complexType>
        </xs:element>

        <xs:element name="eventhandlers_response">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="eventhandler_response" minOccurs="0"
maxOccurs="unbounded" />
                        </xs:sequence>
                        <xs:attribute name="size" type="xs:string" use="required" />
                </xs:complexType>
        </xs:element>

        <xs:element name="eventsubscribe">
                <xs:complexType>
                        <xs:attribute name="type" type="event_type" default="any" />
                        <xs:attribute name="action" type="action_option" default="add" />
                        <xs:attribute name="resource_id" type="xs:string" default="any" />
                        <xs:attribute name="resource_type" type="event_resource_type"
default="any" />
                </xs:complexType>
        </xs:element>

        <xs:element name="conference">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="conf_action" minOccurs="0" maxOccurs="1" />
                        </xs:sequence>
                        <xs:attribute name="type" type="media_type" default="audio" />
                        <xs:attribute name="max_parties" type="xs:string" default="9" />
                        <xs:attribute name="reserve" type="xs:string" default="0" />
                        <xs:attribute name="layout" type="xs:string" />
                        <xs:attribute name="layout_regions" type="xs:string" />
                        <xs:attribute name="layout_size" type="layout_size_option" />
                        <xs:attribute name="caption" type="boolean_type" default="yes" />
                        <xs:attribute name="caption_duration" type="time_value" default="20s" />
                        <xs:attribute name="beep" type="boolean_type" default="yes" />
```

```
                        <xs:attribute name="clamp_dtmf" type="boolean_type" default="yes" />
                        <xs:attribute name="auto_gain_control" type="boolean_type" default="yes"
/>
                        <xs:attribute name="echo_cancellation" type="boolean_type" default="yes"
/>
                        <xs:attribute name="active_talker_region" type="xs:string" />
                        <xs:attribute name="active_talker_interval">
                                <xs:simpleType>
                                        <xs:restriction base="xs:string">
                                                <xs:pattern value="(\+)?([0-9]*\.)?[0-
9]+(ms|s)|infinite|0"/>
                                        </xs:restriction>
                                </xs:simpleType>
                        </xs:attribute>
                        <xs:attribute name="max_active_talkers">
                                <xs:simpleType>
                                        <xs:restriction base="xs:string">
                                                <xs:pattern value="[2-9]|10"/>
                                        </xs:restriction>
                                </xs:simpleType>
                        </xs:attribute>
                </xs:complexType>
        </xs:element>

        <xs:element name="conference_response">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="conf_action" minOccurs="0" maxOccurs="1" />
                                <xs:element ref="conf_participant" minOccurs="0"
maxOccurs="unbounded"/>
                        </xs:sequence>
                        <xs:attribute name="type" type="media_type" />
                        <xs:attribute name="max_parties" type="xs:string"  />
                        <xs:attribute name="reserve" type="xs:string" />
                        <xs:attribute name="layout" type="xs:string"  />
                        <xs:attribute name="layout_regions" type="xs:string" />
                        <xs:attribute name="layout_size" type="layout_size_option" />
                        <xs:attribute name="caption_duration" type="time_value"  />
                        <xs:attribute name="beep" type="xs:string" default="yes" />
                        <xs:attribute name="clamp_dtmf" type="boolean_type"/>
                        <xs:attribute name="auto_gain_control" type="boolean_type"/>
                        <xs:attribute name="echo_cancellation" type="boolean_type"/>
                        <xs:attribute name="active_talker_region" type="xs:string" />
                        <xs:attribute name="active_talker_interval">
                                <xs:simpleType>
                                        <xs:restriction base="xs:string">
                                                <xs:pattern value="(\+)?([0-9]*\.)?[0-
9]+(ms|s)|infinite|0"/>
                                        </xs:restriction>
                                </xs:simpleType>
                        </xs:attribute>
                        <xs:attribute name="max_active_talkers">
                                <xs:simpleType>
                                        <xs:restriction base="xs:string">
                                                <xs:pattern value="[2-9]|10"/>
                                        </xs:restriction>
                                </xs:simpleType>
                        </xs:attribute>
                        <xs:attributeGroup ref="response_attrgroup" />
                </xs:complexType>
        </xs:element>

        <xs:element name="conferences_response">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="conference_response" minOccurs="0"
maxOccurs="unbounded"/>
                        </xs:sequence>
                        <xs:attribute name="size" type="xs:string" use="required" />
                </xs:complexType>
        </xs:element>
```

```
        <xs:element name="mrcp">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="mrcp_action" minOccurs="0" maxOccurs="1" />
                        </xs:sequence>
                        <xs:attribute name="asr" type="boolean_type" default="yes" />
                        <xs:attribute name="tts" type="boolean_type" default="yes" />
                </xs:complexType>
        </xs:element>

        <xs:element name="mrcp_response">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="mrcp_action" minOccurs="0" maxOccurs="1" />
                        </xs:sequence>
                        <xs:attribute name="asr" type="boolean_type" />
                        <xs:attribute name="tts" type="boolean_type" />
                        <xs:attributeGroup ref="response_attrgroup" />
                </xs:complexType>
        </xs:element>

        <xs:element name="mrcps_response">
                <xs:complexType>
                        <xs:sequence>
                                <xs:element ref="mrcp_response" minOccurs="0" maxOccurs="unbounded"
/>
                        </xs:sequence>
                        <xs:attribute name="size" type="xs:string" use="required" />
                </xs:complexType>
        </xs:element>

        <xs:element name="web_service">
                <xs:complexType>
                        <xs:choice minOccurs="1" maxOccurs="1">
                                <xs:element ref="call" />
                                <xs:element ref="call_response" />
                                <xs:element ref="calls_response" />
                                <xs:element ref="conference" />
                                <xs:element ref="conference_response" />
                                <xs:element ref="conferences_response" />
                                <xs:element ref="eventhandler" />
                                <xs:element ref="eventhandler_response" />
                                <xs:element ref="eventhandlers_response" />
                                <xs:element ref="mrcp" />
                                <xs:element ref="mrcp_response" />
                                <xs:element ref="mrcps_response" />
                                <xs:element ref="event" />
                                <xs:element ref="error" />
                        </xs:choice>
                        <xs:attribute name="version" type="xs:NMTOKEN" fixed="1.0" use="required"
/>
                </xs:complexType>
        </xs:element>
</xs:schema>
```

To simplify PowerMedia XMS RESTful application programming in Java, see the following Tech Note:

http://www.dialogic.com/~/media/products/media-server-software/download/xms-demos/XMS-XMLBeans_Technote_20130405.pdf

7. Dynamic Text and Image Generation

In RESTful mode, PowerMedia XMS uses ImageMagick libraries (http://www.imagemagick.org) to enable images and captions to be created dynamically by the application. This simplifies the task of building menus and other static images. Instead of having to create every image in advance and maintain them as the application evolves, a template for image creation is used, and text and images plugged into the template. This feature may also be used to create graphics for simple presentations and games.

Images are built from three data sources:

- A predefined document template, stored on PowerMedia XMS
- A predefined style definition, stored on PowerMedia XMS
- Dynamic content data provided by the application, via the PowerMedia XMS RESTful API, at run-time

PowerMedia XMS combines these to create and display a JPEG image at runtime.

Document Templates

Templates are used to define the structural elements that make up the image.

Template files are stored in the directory: `/etc/xms/imagemaker.d` and have names ending with the extension *.conf*. PowerMedia XMS will load such named files in this directory at start up. These files use the familiar "ini" file format and may contain multiple template definitions. Each section defines a single template with the name of the section serving as a globally unique name.

Note: The PowerMedia XMS Admin Console cannot be used to modify the *.conf* files. You will need to SSH into the system to make any modifications ("root" user with password "powermedia" or whatever you may have changed the root password to).

The following is an example template that defines a basic menu:

```
;-------------------------------------------------------------------------------
; Example ImageMaker document templates.
; This file should be placed into the imagemaker.d directory.
;----------------------------------------------------------------------------
;
; Each section defines one document template.
;
; ---------------------------------------------------------------------------
; | Parameter | Value     | Description                                     |
; |-----------+-----------+-------------------------------------------------|
; |  style    | filename  | File containing style information for the doc.  |
; |-----------+-----------+-------------------------------------------------|
; |  text     | Unique id | A text element                                  |
; |-----------+-----------+-------------------------------------------------|
; |  list     | Unique id | A list element (text)                           |
; |-----------+-----------+-------------------------------------------------|
; |  image    | Unique id | An image element                                |
; ---------------------------------------------------------------------------
;
; The following values are reserved and must not be used for
; element ids: id document
;----------------------------------------------------------------------------
;
```

```
; A basic menu. Three elements: A title, list of options and a footer.
[menu]
style = /var/lib/xms/imagemaker/style/menu.style
text = header
list = items
text = footer
; End of template
```

Style Definition

The appearance of the template's elements is defined using a separate style document, which is located in the directory: `/var/lib/xms/imagemaker/style`. A single style document may be shared by multiple templates. Style definitions for unknown elements are simply ignored. This behavior allows several templates to use a single style document and provide a common theme.

Rather than simply setting a background color for a style, the style file may specify an image file to be used as the background for the image being generated. This allows a common theme or logo to be included in each.

The following is an example of a CIF resolution JPEG background file:

The image generator supports many image formats including PNG. Using the PNG format allows layering effects to be achieved through the use of transparency in background images. All image files used to compose the output image are stored in the directory: `/var/lib/xms/imagemaker/images`.

The following is an example style document used for the menu defined earlier in this section:

Note: All values given in the style file are in pixels. Different resolutions are made up of a different arrays of pixels (CIF=352x288, VGA=640x480 and 720p=1280x720); therefore, a different style file with different values is required for each resolution. In addition, the aspect ratio (height to width proportion) also differs among resolutions and needs to be considered when defining layout.

```
; ---------------------------------------------------------------------------
; Style information for a full-screen menu.
;
; This file consists of a section named 'document' that defines the main
; style for the doc. This followed by sections that specify style information
; for the corresponding element in the document that this style is applied ; ; to.
; Colors may be defined as: a css hex value, a css color name or transparent.
; See http://www.w3.org/TR/SVG/types.html#ColorKeywords for color names.
;
; height             height of document or element in pixels (CIF is 288).
; width              width of document or element in pixels (CIF is 352).
; top                offset of top of an element relative to the document.
; left               offset of left of an element relative to the document.
; padding            offset of the content relative to top and left.
; background-image   filename of an optional background image for the
;                    document.
; background-color   color used to fill the document or element background.
; color              color used for text.
; border-radius      corner radius in pixels.
; border-color       color used for element border.
; border-width       element border width in pixels.
; text-align         justification of the text (left, center, right).
; font-size          size of the text font  (small, medium, large).
;----------------------------------------------------------------------------
[document]
height = 288
width = 352
background-image = /var/lib/xms/imagemaker/images/dialogic_background.png
background-color = #162660
color = #fff
[header]
height = 36
width = 328
top = 12
left = 12
padding = 4
background-color = aliceblue
color = #003
border-radius = 6
text-align = center
font-size = large

[items]
height = 184
width = 328
top = 60
left = 12
padding = 12
background-color = transparent
color = #fff
text-align = left
font-size = medium
```

```
[footer]
height = 20
width = 328
top = 260
left = 12
padding = 2
background-color = transparent
color = #9999a8
text-align = center
font-size = small

; End of style file
; ---------------------------------------------------------------------------
```

Creating the Image using the RESTful API

A PowerMedia XMS RESTful application is able to use the dynamic image creation feature with the following functions:

- play
- playcollect
- playrecord
- overlay

The image contents are specified as a single URI string that is prefixed by "image:", followed by a number of ampersand (&) separated parameters.

Each parameter is defined as a name followed by an equal sign (=) followed by a value. The first parameter uses the reserved name 'id' which specifies the name of the template being used. The remaining parameters specify template element names and their respective contents.

With the exception of the list element type, all elements take only a single value. The list of elements may be defined multiple times in order to define each item.

If a parameter value needs to contain ampersand or equal characters, these must be passed as percent escaped hex values, for example, an ampersand is represented by "%26".

An example of the URI used to create a simple menu is shown below. It contains a header, five (5) items and a footer.

```
uri="image:id=menu&header=Today’s Menu&items=1 Salad&items=2 Chips&items=3 Burger&items=4
Pizza&items=5 Panini&items=7 Pasta&footer=Tuesday"
```

When using XMSTool to drive an XMS application, (refer to XMSTool RESTful Utility) the ampersand (&) needs to be specified as "&". This same URI syntax should then appear as follows:

```
uri="image:id=menu&header=Today’s Menu&items=1 Salad&amp;items=2 Chips&items=3 Burger&items=4
Pizza&amp;items=5 Panini&items=7 Pasta&footer=Tuesday"
```

Overlay and the play/playcollect/playrecord commands are different in that the overlay command is used to superimpose the image on an existing media stream such as a mixed conference. With play/playcollect/playrecord, the defined image itself is used as basis for the video stream.

Note: Currently, only a single URI can be played for both audio and video. If an "image" URI is played, then any audio meant to accompany it will be lost. This restriction is planned to be lifted in a future release.

Combining the Three Data Sources

Combining the template file, style file, background image, and dynamic text representation results in the following menu:

There are several examples of different images included with PowerMedia XMS. Start with a working image and experiment/adjust until the desired results are obtained.

8. XMSTool RESTful Utility

This section provides details about the XMSTool RESTful Utility (also referred to herein as "XMSTool" or "Utility"). XMSTool is used for developing, debugging, and supporting applications for the PowerMedia XMS using the HTTP RESTful API.

XMSTool is a java-based test application for passing and receiving XMS RESTful API messages to and from the PowerMedia XMS. It can be used to build and parse individual RESTful messages, and can drive and record simple applications.

The utility provides the following:

- Support for both 1PCC and 3PCC (see the Call Control Models)
- Ability to manually enter and execute the XMS RESTful API commands and observe the results
- Method to record Macros for automated execution of command sequences (Demo mode), enabling users to create simple Demos and debug their applications
- Pre-recorded Macros available for commonly used call scenarios
- Logging capabilities

XMSTool can be run in two different modes:

- Demo/Simple Mode
  Uses predefined XML scripts; short application scenarios can be executed to demonstrate most of the PowerMedia XMS RESTful functionality. Session logging is available to examine the message interchange. Only sessions using inbound SIP calls are currently available in this mode.
- Advanced Mode
  Allows individual RESTful commands to be manually entered for full XMS control. This mode is intended to be used by developers who are looking to become familiar with the RESTful API messages used to control XMS. It also allows the individual commands that make up a Macro/Demo to be recorded for replay or to provide an accurate way to reproduce a problem in PowerMedia XMS.

For detailed information about using XMSTool, refer to the *Dialogic® PowerMedia™ XMS Installation and Configuration Guide*.