



# **Dialogic® PowerMedia™ IP Media Server**

## **Release 3.0.0**

Application Developer's Guide



---

# Copyright and Legal Notice

Copyright © 2000-2010 Dialogic Corporation. All Rights Reserved. You may not reproduce this document in whole or in part without permission in writing from Dialogic Corporation at the address provided below.

All contents of this document are furnished for informational use only and are subject to change without notice and do not represent a commitment on the part of Dialogic Corporation or its subsidiaries ("Dialogic"). Reasonable effort is made to ensure the accuracy of the information contained in the document. However, Dialogic does not warrant the accuracy of this information and cannot accept responsibility for errors, inaccuracies or omissions that may be contained in this document.

INFORMATION IN THIS DOCUMENT IS PROVIDED IN CONNECTION WITH DIALOGIC® PRODUCTS. NO LICENSE, EXPRESS OR IMPLIED, BY ESTOPPEL OR OTHERWISE, TO ANY INTELLECTUAL PROPERTY RIGHTS IS GRANTED BY THIS DOCUMENT. EXCEPT AS PROVIDED IN A SIGNED AGREEMENT BETWEEN YOU AND DIALOGIC, DIALOGIC ASSUMES NO LIABILITY WHATSOEVER, AND DIALOGIC DISCLAIMS ANY EXPRESS OR IMPLIED WARRANTY, RELATING TO SALE AND/OR USE OF DIALOGIC PRODUCTS INCLUDING LIABILITY OR WARRANTIES RELATING TO FITNESS FOR A PARTICULAR PURPOSE, MERCHANTABILITY, OR INFRINGEMENT OF ANY INTELLECTUAL PROPERTY RIGHT OF A THIRD PARTY.

Dialogic products are not intended for use in medical, life saving, life sustaining, critical control or safety systems, or in nuclear facility applications.

Due to differing national regulations and approval requirements, certain Dialogic products may be suitable for use only in specific countries, and thus may not function properly in other countries. You are responsible for ensuring that your use of such products occurs only in the countries where such use is suitable. For information on specific products, contact Dialogic Corporation at the address indicated below or on the web at [www.dialogic.com](http://www.dialogic.com).

It is possible that the use or implementation of any one of the concepts, applications, or ideas described in this document, in marketing collateral produced by or on web pages maintained by Dialogic may infringe one or more patents or other intellectual property rights owned by third parties. Dialogic does not provide any intellectual property licenses with the sale of Dialogic products other than a license to use such product in accordance with intellectual property owned or validly licensed by Dialogic and no such licenses are provided except pursuant to a signed agreement with Dialogic. More detailed information about such intellectual property is available from Dialogic's legal department at 9800 Cavendish Blvd., 5th Floor, Montreal, Quebec, Canada H4M 2V9. **Dialogic encourages all users of its products to procure all necessary intellectual property licenses required to implement any concepts or applications and does not condone or encourage any intellectual property infringement and disclaims any responsibility related thereto. These intellectual property licenses**

---

**may differ from country to country and it is the responsibility of those who develop the concepts or applications to be aware of and comply with different national license requirements.**

Any use case(s) shown and/or described herein represent one or more examples of the various ways, scenarios or environments in which Dialogic® products can be used. Such use case(s) are non-limiting and do not represent recommendations of Dialogic as to whether or how to use Dialogic products.

Dialogic, Dialogic Pro, Brooktrout, Diva, Diva ISDN, Making Innovation Thrive, Video is the New Voice, Diastar, Cantata, TruFax, SwitchKit, SnowShore, Eicon, Eicon Networks, NMS Communications, NMS (stylized), Eiconcard, SIPcontrol, TrustedVideo, Exnet, EXS, Connecting to Growth, Fusion, Vision, PacketMedia, NaturalAccess, NaturalCallControl, NaturalConference, NaturalFax and Shiva, among others as well as related logos, are either registered trademarks or trademarks of Dialogic Corporation or its subsidiaries. Dialogic's trademarks may be used publicly only with permission from Dialogic. Such permission may only be granted by Dialogic's legal department at 9800 Cavendish Blvd., 5th Floor, Montreal, Quebec, Canada H4M 2V9. Any authorized use of Dialogic's trademarks will be subject to full respect of the trademark guidelines published by Dialogic from time to time and any use of Dialogic's trademarks requires proper acknowledgement.

Microsoft is a registered trademark of Microsoft Corporation in the United States and/or other countries. Other names of actual companies and products mentioned herein are the trademarks of their respective owners.

This document discusses one or more open source products, systems and/or releases. Dialogic is not responsible for your decision to use open source in connection with Dialogic products (including without limitation those referred to herein), nor is Dialogic responsible for any present or future effects such usage might have, including without limitation effects on your products, your business, or your intellectual property rights.

---

## Hardware Limited Warranty

Please refer to the following Dialogic web site for information on hardware warranty information, which applies unless different terms have been agreed to in a signed agreement between yourself and Dialogic Corporation or its subsidiaries. The listed hardware warranty periods and terms are subject to change without notice. For purchases not made directly from Dialogic please contact your direct vendor in connection with the warranty period and terms that they offer.

<http://www.dialogic.com/warranties>

# Contents

<b>About this Publication</b> .....	<b>23</b>
Using this Publication .....	24
Audience and Purpose .....	24
Documentation Set .....	25
Document Conventions .....	25
Notes, Cautions, and Warnings .....	25
Links in PDF .....	26
Contacting Dialogic Technical Services and Support .....	27
Ordering Licenses .....	27
<b>1 - Introduction</b> .....	<b>28</b>
Control Protocols and Services .....	29
Call Control Protocols .....	30
Session Initiation Protocol (SIP) .....	31
Session .....	31
Service Indicators .....	31
Session Description Protocol (SDP) .....	32
Media Server Control Markup Language (MSCML) .....	32
SIP Methods with MSCML .....	32
VoiceXML .....	33
Media Storage, Processing, and Supported Codecs .....	35
Audio .....	36
Audio Content Storage .....	36
Video .....	37
Mixed Audio/Video .....	37
WAV-based .....	37
3GPP and 3GPP2 .....	37
Mixed Audio/Video Content Storage .....	38
File Storage and Retrieval .....	38
Network File System (NFS) .....	38

HTTP .....	38
Real Time Streaming Protocol (RTSP) .....	38
DTMF .....	39
In-Band Busy Tone Detection and Reporting .....	39
Services .....	40
Network Announcements .....	40
Simple Announcements .....	41
Announcement Sequences and Variable Content Announcements .....	41
Conferences .....	42
Simple Conferences .....	42
Advanced Conferences .....	42
Interactive Voice Response (IVR) .....	43
Dialog (VoiceXML) .....	43
Advanced Video Features .....	44
Video Transcoding .....	45
Video Transcoder Settings .....	45
Video Transcoder Automatic Setting Rules .....	46
Video Transcoder Restrictions .....	46
Text overlay .....	47
Real Time Streaming Protocol (RTSP) .....	48
Configuring Session Keep Alive .....	48

## **2 - Session Initiation Protocol (SIP) .....**

Conformance .....	51
Media Server Availability .....	52
Example .....	52
Request .....	52
Response .....	53
SIP Description .....	55
SIP User Agents .....	55
SIP Message .....	55
SIP Transaction .....	56
URLs .....	56
Service Indicators .....	56
SIP Message Body Types .....	57
SIP Requests .....	58
Request Line .....	58
Headers .....	59
Session Descriptor .....	59
SIP Methods .....	60
INVITE .....	60
Headers in an INVITE .....	61
ACK .....	62
Headers in an ACK .....	62
CANCEL .....	63
Headers in CANCEL .....	63
OPTIONS .....	64
Headers in OPTIONS .....	64
BYE .....	65
Headers in BYE .....	65

INFO	66
Headers in INFO	66
PRACK	67
Headers in a PRACK	67
SIP Headers	68
Format and Syntax	69
Supported Headers	69
Accept	69
Call-ID (i)	70
Contact (m)	70
Content-Length (l)	70
Content-Type (c)	71
CSeq	71
From (f)	71
Max-Forwards	72
Record-Route	72
Require	72
Route	72
Session-Expires	72
Server	73
Subject	73
Supported	73
To (t)	73
Unsupported	74
User-Agent	74
Via (v)	74
SIP Responses	75
SIP Provisional Response Configuration	75
SIP Return Codes	75
Session Description Protocol	79
Session Description Headers	79
Time Description Headers	81
Media Description Headers	81
Header Action Classes	82
Header Definitions	82
m= (Media information)	82
c= (Connection data)	83
a= (Attribute Lines)	84
Ports	85
SIP	85
RTP	85
Reliability of Provisional Responses	86
Syntax and Escaping	88
Syntax and MSCML Body	90
IP Media Server Behavior When Hold Media is Presented	91
About Hold SDP	91
Hold Behavior for the Various Media Services	92
<b>3 - Announcement Service API</b>	<b>94</b>
Overview	95



File Retrieval .....	95
Network File System (NFS) .....	95
HTTP .....	95
Announcement Types .....	95
Announcement Service Indicator and Request URI .....	96
Simple Announcements .....	97
Announcement Sequences .....	98
Variable-Content Announcements .....	98
Implementation .....	98
SIP Request Parameters for Announcements .....	100
Variable Types and Subtypes .....	102
<b>4 - Conferencing API .....</b>	<b>106</b>
Simple Conferencing .....	108
How SIP Manages Conferences .....	108
Creating a Simple Conference .....	108
Adding a Participant to a Simple Conference .....	108
Ending a Simple Conference .....	108
Attributes for Simple Conferences and Participants .....	109
DTMF Clamping .....	109
Managing Video Switching .....	110
Call Flow and Sample Code Examples .....	111
Code for Creating a Simple Conference .....	112
Advanced Conferencing .....	114
Using MSCML for Advanced Conferencing .....	114
configure_conference .....	114
configure_leg .....	114
MSCML Attributes and Elements for <configure_conference> .....	115
MSCML Attributes and Elements for <configure_leg> .....	116
MSCML Attributes for <managecontent> .....	117
<managecontent> Examples .....	118
Creating an Advanced Conference .....	119
Modifying an Advanced Conference .....	120
Ending an Advanced Conference .....	121
Joining Participants (Legs) to an Advanced Conference .....	121
Modifying a Conference Participant .....	122
Removing Participants From a Conference .....	122
Conference Subsetting .....	123
Active Talker Events .....	123
IVR Operations during a Conference .....	124
Playing and Recording Within the Entire Conference .....	124
Playing to the Conference .....	125
Recording the Conference Output .....	125
Video Conferencing Enhancements .....	126
MSCML Changes .....	126
IVR Operations on Participant Legs .....	126
Detecting DTMF Digits On A Conference Leg .....	127
Playing Audio to a Participant Leg .....	129
Detecting and Reporting Busy Call Progress Tones in MSCML .....	130
Simultaneous Play and Record .....	133

Creating an Internal Conference Leg . . . . .	133
Recording a Conference . . . . .	133
MSCML Conferencing Requests . . . . .	134
Conferencing Request Elements and Attributes . . . . .	134
configure_conference . . . . .	134
In INFO Message . . . . .	135
configure_leg . . . . .	136
Coached Conferencing . . . . .	138
Overview . . . . .	138
MSCML Elements and Attributes of Coached Conferencing . . . . .	139
configure_team . . . . .	139
Configuring a Coached Conference . . . . .	140
Creating the Conference . . . . .	140
Joining and Configuring the Coach . . . . .	141
Joining and Configuring the Agent . . . . .	141
Joining and Configuring the Client . . . . .	142
Supervisor Query for Number of Team Members . . . . .	143
Exiting the Conference . . . . .	143
Using SIP INFO . . . . .	144
MSCML Conferencing Reference . . . . .	145
MSCML Elements . . . . .	145
activetalkers . . . . .	145
configure_team . . . . .	145
dtmf . . . . .	145
events . . . . .	146
keypress . . . . .	146
notification . . . . .	146
signal . . . . .	146
subscribe . . . . .	146
teammate . . . . .	147
MSCML Attributes . . . . .	147
action . . . . .	147
dtmfclamp . . . . .	147
id . . . . .	148
mixmode . . . . .	148
repeat . . . . .	149
report . . . . .	150
reserveconfmedia . . . . .	150
reservedtalkers . . . . .	151
toneclamp . . . . .	151
type . . . . .	151
<b>5 - IVR with MSCML . . . . .</b>	<b>153</b>
IVR Service . . . . .	154
Call Progress Tones . . . . .	155
Playing Announcements . . . . .	157
Elements and Attributes . . . . .	157
Responses . . . . .	157
Collecting DTMF Digits . . . . .	158
Prompting . . . . .	158
Digit Buffering . . . . .	158

---

Star and Pound Keys .....	159
Timing Attributes .....	159
Responses .....	160
Recording Audio .....	161
Playrecord Attributes .....	161
Playrecord Process .....	161
Timing Attributes .....	162
Additional Attributes .....	162
Responses .....	163
Handling of Content Retrieval Errors .....	163
Stopping an IVR Request in Progress .....	164
ID Attribute .....	164
Response .....	164
MSCML IVR Reference .....	165
IVR Elements .....	165
IVR Prompt Block .....	166
Prompt Elements .....	167
IVR Attributes .....	168
barge .....	168
baseurl .....	168
beep .....	168
cleardigits .....	168
delay .....	169
duration .....	169
encoding .....	169
endsilence .....	169
endwarning .....	170
escapekey .....	170
extradigittimer .....	170
firstdigittimer .....	170
id .....	171
interdigittimer .....	171
initsilence .....	171
locale .....	171
maskdigits .....	171
maxdigits .....	173
mode .....	173
offset .....	173
promptencoding .....	174
recstopmask .....	174
recurl .....	174
recencoding .....	174
repeat .....	175
report .....	175
returnkey .....	176
stop_on_error .....	176
subtype .....	176
type .....	177
url .....	178
value .....	178
warnurl .....	178
IVR Response Elements and Attributes .....	178

Response Elements	179
error_info	179
id	179
Response Attributes	180
code	180
digits	180
playduration	180
reason	180
reclength	181
text	181
<b>6 - VoiceXML Version 2.0 and Dialog Service</b>	<b>182</b>
About VoiceXML	183
VoiceXML Interpreter	183
Dialog Service Indicator and Request URI	183
VoiceXML Launcher	184
Default Script	184
VoiceXML Concepts	185
Syntax	185
Scope	185
Resource Fetching	185
VoiceXML Application and Its Documents	185
Dialogs	186
Scripting	186
DTMF Grammars Guidelines	187
<grammar>	187
<tag>	187
<ruleref>	188
<rule>	188
MRCP	188
Session Variables	188
File Storage and Retrieval	189
Media Content Recovery Extension	189
VoiceXML Elements Reference	191
<assign>	191
<audio>	191
<block>	192
<catch>	192
<choice>	193
<clear>	194
<content>	195
<data>	196
<disconnect>	197
<else>	197
<elseif>	197
<enumerate>	198
<error>	198
<exit>	198
<field>	199
<filled>	200
<font>	201

<form>	202
<goto>	202
<grammar>	203
<help>	205
<if>	205
<initial>	206
<link>	206
<log>	208
<menu>	208
<meta>	209
<noinput>	209
<nomatch>	210
<object>	210
<option>	211
<overlay>	211
<param>	213
<prompt>	213
<property>	214
<record>	215
<reprompt>	216
<return>	216
<say-as>	217
<script>	217
<scroll>	218
<subdialog>	219
<submit>	220
<throw>	221
<transfer>	222
<value>	225
<var>	225
<vxml>	226
VoiceXML Properties	227
com.snowshore.criticaldigit_timer	227
MIME Types Supported	228
ECMAScript Functionality	230
VoiceXML Extended Session Variables	233
VoiceXML 2.0 Recommendations	233
Dialogic® Extensions	233
Example	234
Text Overlay Functionality	236
Text Overlay Overview	236
Characteristics of Text Overlay	236
Text Overlay Restrictions	237
Installing Font Files	237
Dynamic Fonts	237
Cached Fonts	238
trcr.cfg Description	238
Sample trcr.cfg	240
Text Overlay Demos	240
<b>7 - Sample Code and Call Flows</b>	<b>241</b>

Announcements . . . . .	242
Play an Announcement as Early Media . . . . .	242
Call Flow for an Early Media Announcement . . . . .	244
Playing an Announcement as Normal Media . . . . .	245
Call Flow for a Normal Media Announcement . . . . .	247
Stopping Media—Hold . . . . .	248
Call Flow for Stopping Media—Hold . . . . .	250
Repeating the Audio . . . . .	251
Conferences . . . . .	252
Creating a Simple Conference . . . . .	252
Call Flow for a Simple Conference (Normal Media) . . . . .	253
Creating an Advanced Conference . . . . .	254
Call Flow to Set up an Advanced Conference . . . . .	256
Modifying Conference Using Subscribe . . . . .	257
Providing Communication for Participant in an Advanced Conference . . . . .	258
Joining a Participant Using Special Attributes . . . . .	259
Suspending Communications within a Conference . . . . .	260
Response to Mute a Conference Participant . . . . .	261
Playing Audio to Conference Participant . . . . .	262
Changing Mixmode to Parked . . . . .	262
Response to Parked . . . . .	263
Playing the Audio . . . . .	263
Response to Message to Play Audio . . . . .	264
Changing Participant Mixmode Back to Full . . . . .	264
Response to Mixmode Change . . . . .	265
Playcollect and Playrecord in a Conference . . . . .	266
Changing Participant Mixmode to Parked . . . . .	266
Response to Parked . . . . .	266
PlayCollect . . . . .	267
Response to PlayCollect . . . . .	267
Sending PlayRecord . . . . .	268
Response to PlayRecord . . . . .	268
Changing Participant Back to Full . . . . .	269
Response to Full . . . . .	269
Coached Conferencing . . . . .	270
IVR with MSCML . . . . .	282
Playing a Simple Announcement . . . . .	282
Play Payload . . . . .	282
Expected Response . . . . .	282
Playing a Sequenced Announcement . . . . .	282
Play Payload . . . . .	282
Stopping a Play Command . . . . .	283
Request 1 Payload . . . . .	283
Request 2 Payload . . . . .	283
Expected Response to Request 1 Payload . . . . .	283
Expected Response to Request 2 Payload . . . . .	283
PlayCollect . . . . .	284
Payload . . . . .	284
Expected Response . . . . .	284
Playing a Recording . . . . .	284
Payload . . . . .	284

---

Expected Response .....	284
Stopping a Recording .....	285
Request 1 Payload .....	285
Request 2 Payload .....	285
Expected Response Request 1 Payload .....	285
Expected Response Request 2 Payload .....	285
Asynchronous DTMF .....	286
Subscribing to Standard Digit Events .....	286
Subscribing to Long Digit Events .....	286
Subscribing to Both Standard and Long Digit Events .....	286
Turning Off Digit Event Reporting .....	287
Example Responses .....	287
Call Flow for IVR with MSCML .....	289
Call Flow for PIN Collection, IVR with MCSCML .....	290
Explanation of Call Flow .....	290
Call Flow for Recording a Message, IVR with MSCML .....	292
Explanation of Call Flow .....	293
VoiceXML .....	295
Playing an Announcement .....	295
PIN Collection .....	295
Call Flow for VoiceXML .....	297
Transferring a Call .....	298
Using VCR and Gain Controls .....	300
Text Overlay .....	304
Simple Text Overlay .....	304
Scrolling Text Overlay .....	304
<b>A - Audio Library .....</b>	<b>305</b>
Sound Library .....	306
Phrases and Messages .....	306
Numbers .....	309
Dates and Ordinal Numbers .....	310
Letters .....	311
Time and Money .....	312
Press Keys .....	313
Quantities .....	314
Miscellaneous Words .....	315
Generic Audio Files .....	317
<b>B - VoiceXML Version 1.0 and Dialog Service .....</b>	<b>319</b>
About VoiceXML .....	321
VoiceXML Interpreter .....	321
Dialog Service Indicator and Request URI .....	321
VoiceXML Launcher .....	322
VoiceXML Concepts .....	322
Syntax .....	322
Scope .....	322
Resource Fetching .....	323
VoiceXML Application and Its Documents .....	323

---

Dialogs .....	323
Grammar and Scripting .....	324
Session Variables .....	324
File Storage and Retrieval .....	324
Media Content Recovery Extension .....	324
VoiceXML Elements Reference .....	326
<assign> .....	326
<audio> .....	326
<block> .....	327
<break> .....	327
<catch> .....	328
<choice> .....	328
<clear> .....	329
<data> .....	329
<disconnect> .....	330
<dtmf> .....	330
<else> .....	331
<elseif> .....	331
<error> .....	331
<exit> .....	332
<field> .....	332
<filled> .....	333
<form> .....	333
<goto> .....	334
<grammar> .....	334
<help> .....	334
<if> .....	335
<initial> .....	335
<link> .....	336
<log> .....	336
<menu> .....	337
<meta> .....	337
<noinput> .....	338
<nomatch> .....	338
<param> .....	339
<prompt> .....	339
<property> .....	340
<record> .....	340
<reprompt> .....	341
<return> .....	341
<sayas> .....	341
<script> .....	342
<subdialog> .....	342
<submit> .....	343
<throw> .....	344
<transfer> .....	344
<value> .....	346
<var> .....	346
<vxml> .....	347
VoiceXML Attributes Reference .....	348
application .....	348
bargein .....	348



---

bargeintype	348
base	348
beep	349
bridge	349
class	349
cond	350
connect-timeout	350
content	350
count	350
dest	351
destexpr	351
dtmf	351
dtmfterm	351
event	352
expr	352
expterm	352
finalsilence	352
http-equiv	352
id	352
longdigit	353
max-time	353
method	353
modal	353
mode	354
msecs	354
name	354
namelist	354
next	355
nextitem	355
recsrc	355
reqUri	355
scope	355
size	355
slot	356
src	356
srcexpr	356
stopdigits	356
timeout	357
transfer-audio	357
type	357
value	357
valuetype	358
version	358
VCR	358
video	358
VoiceXML Properties	359
com.snowshore.criticaldigit_timer	359
MIME Recording Encoding Types	359
ECMAScript Functionality	361
Support for VoiceXML Extended Session Variables	364
VoiceXML 2.0 Recommendations	364
Dialogic® Extensions	364

---

Example .....	365
<b>C - MSCML Schema</b> .....	<b>367</b>
<b>D - RTP Codec Selection with VoiceXML &lt;transfer/&gt;</b> .....	<b>378</b>
<b>E - Using AMR-NB</b> .....	<b>380</b>
Using AMR-NB .....	381
Media Description Header .....	381
Dynamic Payload Type .....	381
Format Specific Parameters .....	382
Optional Format Parameters .....	385
Packet Time .....	385
Maximum Packet Time .....	385
<b>F - Dial Pulse Detection</b> .....	<b>386</b>
Overview .....	387
Dial Pulse and DTMF .....	387
Consistency .....	387
MSCML and VXML 1.0/2.0 Support .....	387
Configuration .....	388
Parameters .....	388
Description of Dial Pulse Algorithm .....	390
<b>G - MSCML MRCP Session Management</b> .....	<b>391</b>
Overview of MRCP Session Management .....	392
Features Enabled .....	392
Process .....	392
MSCML Requests .....	395
Create session request format .....	395
Attributes .....	396
Create session response format .....	396
Terminate session request format .....	397
Terminate session response format .....	398
Sample Call Flow .....	399
Scenario .....	399
Call Flow .....	399
<b>H - T.30 Fax Detection</b> .....	<b>405</b>
Fax Call Transfer Call Flow .....	406
Description .....	407
CNG Tone Detection and Event Notification .....	408
VoiceXML Implementation .....	408
VoiceXML Properties .....	409
com.snowshore.signal .....	409
Fax Detection—Example Script .....	410

---

Fax Call Transfer .....	412
Call Transfer—Example Script .....	412
<b>I - T.38 Fax Detection, Termination, and Initiation .....</b>	<b>415</b>
Detection and Termination .....	415
Initiation (Email to Fax) .....	415
Snowshore.cfg Parameters .....	417
Fax Call Termination Call Flow .....	418
Description .....	419
CNG Tone Detection and Event Notification .....	420
VoiceXML 1.0 Implementation .....	420
Fax Detection—Example VXML Script .....	421
VXML Fax Record .....	422
Call Record - Example VXML Script .....	422
<b>Index .....</b>	<b>423</b>

# List of Figures

Figure 1.	Call Control Architecture	30
Figure 2.	Network Announcements	40
Figure 3.	Role of the IP Media Server with RTSP Server	48
Figure 4.	Call Flow: SIP Provisional Response	88
Figure 5.	Playing an Announcement Sequence	99
Figure 6.	Call Flow for Simple Conference (Normal Media)	112
Figure 7.	Sharing of Root Document in VoiceXML Application	186
Figure 8.	Call Flow: Announcement, Early Media	244
Figure 9.	Call Flow: Announcement, Normal Media	247
Figure 10.	Call Flow: Stopping Media Using Hold	250
Figure 11.	Call Flow: Simple Conference	253
Figure 12.	Call Flow: Advanced Conference	256
Figure 13.	Call Flow: IVR with MSCML	289
Figure 14.	Call Flow: PIN Collection, IVR with MSCML	290
Figure 15.	Call Flow: Recording, IVR with MSCML	292
Figure 16.	Call Flow: VoiceXML	297
Figure 17.	Sharing of Root Document in VoiceXML Application	323
Figure 18.	Dial Pulse Parameters	390
Figure 19.	MRCP Session Management	393
Figure 20.	Fax Call Transfer	406
Figure 21.	Fax Call Transfer	418

# List of Tables

Table 1.	SIP Service Indicators . . . . .	31
Table 2.	MSCML Conferencing and MSCML IVR Requests . . . . .	32
Table 3.	Functionality Supported by VoiceXML 1.0 and VoiceXML 2.0 . . . . .	34
Table 4.	Supported Audio Codecs . . . . .	35
Table 5.	Supported Video Codecs . . . . .	35
Table 6.	Audio Encoding and File Storage Formats . . . . .	36
Table 7.	Multimedia Encoding and File Storage Formats . . . . .	38
Table 8.	Application Service Indicators . . . . .	56
Table 9.	Valid Content (MIME) Types . . . . .	57
Table 10.	Supported SIP Methods . . . . .	60
Table 11.	SIP Headers in an INVITE . . . . .	61
Table 12.	SIP Headers in an ACK Method . . . . .	62
Table 13.	SIP Headers in CANCEL Method . . . . .	63
Table 14.	OPTIONS Method Headers . . . . .	64
Table 15.	BYE Method Headers . . . . .	65
Table 16.	INFO Method Headers . . . . .	66
Table 17.	Headers in PRACK Method . . . . .	67
Table 18.	Supported SIP Headers . . . . .	68
Table 19.	SIP Return Codes Generated by the IP Media Server . . . . .	76
Table 20.	SDP Description Headers . . . . .	79
Table 21.	SDP Time Description Headers . . . . .	81
Table 22.	SDP Media Description Headers . . . . .	81
Table 23.	Action Classes . . . . .	82
Table 24.	Unreserved Characters in SIP Request . . . . .	89
Table 25.	Unreserved Characters in User Portion . . . . .	89
Table 26.	Unreserved Characters for Parameters . . . . .	90
Table 27.	Simple Conference Attributes . . . . .	109
Table 28.	Advanced Conferencing Attributes . . . . .	115
Table 29.	Subscribe Element . . . . .	115

---

Table 30.	Attributes for Participant Legs in Advanced Conference . . . . .	116
Table 31.	Subscribe Element for Participant Legs . . . . .	116
Table 32.	Attributes for <managecontent> in an Advanced Conference . . . . .	117
Table 33.	Configure_Leg Attributes . . . . .	121
Table 34.	Configure_Conference Attributes in INVITE Message. . . . .	134
Table 35.	Subscribe Element in INVITE with Configure_Conference . . . . .	135
Table 36.	Configure_Conference Attributes in INFO Message . . . . .	135
Table 37.	Subscribe Element in INFO with Configure_Conference . . . . .	136
Table 38.	Configure_Leg Attributes . . . . .	136
Table 39.	Subscribe Element for Configure_Leg . . . . .	137
Table 40.	Configure_Team Attributes. . . . .	139
Table 41.	MSCML Elements and Attributes for Creating Coached Conferencing . . . . .	140
Table 42.	Timing Attributes . . . . .	162
Table 43.	Additional Attributes of PlayRecord . . . . .	162
Table 44.	MSCML Elements for IVR . . . . .	165
Table 45.	Supported MIME Types, Formats, and Audio Encodings. . . . .	228
Table 46.	Audio Codec Parameter Values. . . . .	228
Table 47.	Supported MIME Types, Formats, and Video Encodings. . . . .	229
Table 48.	ECMAScript Functionality. . . . .	230
Table 49.	VCR and Gain Controls Support for RTSP in VoiceXML 2.0 . . . . .	300
Table 50.	Prompt Library: Standard Phrases. . . . .	307
Table 51.	Prompt Library: Cardinal Numbers. . . . .	310
Table 52.	Prompt Library: Dates and Ordinal Numbers . . . . .	310
Table 53.	Prompt Library: Time and Money Phrases. . . . .	312
Table 54.	Prompt Library: Press Key Phrases. . . . .	313
Table 55.	Prompt Library: Quantities . . . . .	314
Table 56.	Prompt Library: Miscellaneous Words . . . . .	315
Table 57.	Generic Prompt Phrases . . . . .	317
Table 58.	Supported MIME Types, Formats, and Audio Encodings. . . . .	359
Table 59.	Accepted WAVE Format Specifiers . . . . .	360
Table 60.	Codec Parameter Values . . . . .	360
Table 61.	ECMAScript Functionality. . . . .	361
Table 62.	SDP Offer Sent in Outbound SIP INVITE. . . . .	379
Table 63.	Dial Pulse Parameters . . . . .	388
Table 64.	Description of MRCP Session Management Topology. . . . .	394
Table 65.	Attributes of mrcp_create_session. . . . .	396
Table 66.	Attributes of mrcp_session_response . . . . .	397
Table 67.	Attributes of mrcp_terminate_session_request . . . . .	398
Table 68.	T.38 Fax Parameters . . . . .	417

# About this Publication

The Dialogic® PowerMedia™ IP Media Server (which is also referred to herein as "IP Media Server", "IPMS", or "Media Server") is a standards-based SIP, VoiceXML, and MSCML server that performs a wide variety of media processing functions.

The IP Media Server is also an economical and scalable IP media option, as it can power a broad range of voice and video services for next generation wireline, wireless, and broadband services.

This section covers the following topics:

- ◆ [Using this Publication](#)
- ◆ [Documentation Set](#)
- ◆ [Contacting Dialogic Technical Services and Support](#)

# Using this Publication

## Audience and Purpose

This manual is for application developers who choose to use or employ the IP Media Server to deploy network announcements, conferences, Interactive Voice Response (IVR) and more in a voice over IP environment. Appendix A, [“Audio Library”](#), describes the IP Media Server’s preconfigured sound library (audio files), which consists of phrases and messages, numbers, time and money, quantities, and miscellaneous words.



---

# Documentation Set

Dialogic provides the following publications for the Dialogic® IP Media Server:

- ◆ The *Software Quick Start Guide* describes how to install and configure Red Hat Enterprise Linux and IP Media Server software, configure a softphone, and run a demo.
- ◆ The *Installation and Operations Guide* provides instructions for configuring, administering, and maintaining the IP Media Server.
- ◆ The *Application Developer's Guide* provides information for application developers who choose to use the IP Media Server to deploy network announcements, conferences, and Interactive Voice Response (IVR) in a voice over IP (VoIP) environment.
- ◆ The *Command Line Interface Reference Guide* describes the CLI utility which can be used to configure and troubleshoot the IP Media Server.
- ◆ *Installing Red Hat Enterprise Linux 5.0 for the IP Media Server* describes how to install and configure Red Hat Enterprise Linux 5 if you are installing the licensed software version of the IP Media Server.
- ◆ The *License Activation Guide* describes how to activate the license for your IP Media Server.
- ◆ *Upgrading from Release 2.6.0 to 3.0.0 on Red Hat Enterprise Linux Platform* provides information and instructions for upgrading from IP Media Server Release 2.6.0 to IP Media Server Release 3.0.0 on platforms running Red Hat Enterprise Linux Platform. It also includes instructions for downgrading in the event that you need to restore your previous configuration.

These publications, as well as Release Notes, are available in PDF format at <http://www.dialogic.com/manuals>.

## Document Conventions

Conventions used in this document are described here.

## Notes, Cautions, and Warnings

Notes contain information of general interest.

Cautions and warnings appear when appropriate throughout the manual.

Cautions alert you to situations that can make system administration less effective or compromise system performance or security. For example:



---

**Before changing the configuration of a running system, always back up the current configuration using the System→Config command.**

---

Warnings alert you to situations that could cause physical harm to an operator or damage to the IP Media Server. For example:



---

**If an interface is deactivated, all traffic on that interface will be dropped.**

---

### Links in PDF

Hypertext links in the PDF version of this manual are blue. You can click on a cross-reference link to move to the information it references.

Index entries and Table of Contents listings are also clickable links in the PDF format. After you jump to a link, use the Back button on the Acrobat Reader toolbar to return to your prior location.

# Contacting Dialogic Technical Services and Support

For more information, refer to the Dialogic Technical Services and Support site:

<http://www.dialogic.com/support/>

When reporting an issue to Technical Services and Support, be prepared to provide the following information:

- ◆ Full description of the issue.
- ◆ Version of the IP Media Server software you are using.
- ◆ IP Media Server log files.
- ◆ Whether the issue is reproducible; the steps that you took.

Please note that the latest software update and release notes are available from the Dialogic support page.

## Ordering Licenses

You must have a software license to use the Dialogic® IP Media Server. For directions on how to acquire licenses, see the Dialogic® IP Media Server *License Activation Guide*.

# 1 - Introduction

This chapter provides an overview of the features and functionality of the Dialogic® PowerMedia™ IP Media Server (also referred to in this document as “IP Media Server”, “IPMS”, or “Media Server”).

This chapter includes the following sections:

- ◆ [Control Protocols and Services](#)
- ◆ [Call Control Protocols](#)
- ◆ [Media Storage, Processing, and Supported Codecs](#)
- ◆ [Services](#)
- ◆ [Advanced Video Features](#)
- ◆ [Video Transcoding](#)
- ◆ [Text overlay](#)
- ◆ [Real Time Streaming Protocol \(RTSP\)](#)

## Control Protocols and Services

This section introduces the Session Initiation Protocol (SIP) and XML-based APIs that allow the IP Media Server to perform in an IP network, explains the call-related resources, and defines the supported application services.

The IP Media Server can perform a specialized role in the IP network by providing high-capacity, real-time packet processing for network announcements, conferencing, and Interactive Voice Response (IVR) functions. The IP Media Server can also provide video solutions such as Interactive Voice and Video Response (IVVR), multimedia ringback tone (MRBT), video portal, video mail, and video advertisements.

This section contains the following major topics:

- ◆ [Call Control Protocols](#)
- ◆ [Media Storage, Processing, and Supported Codecs](#)
- ◆ [Services](#)

# Call Control Protocols

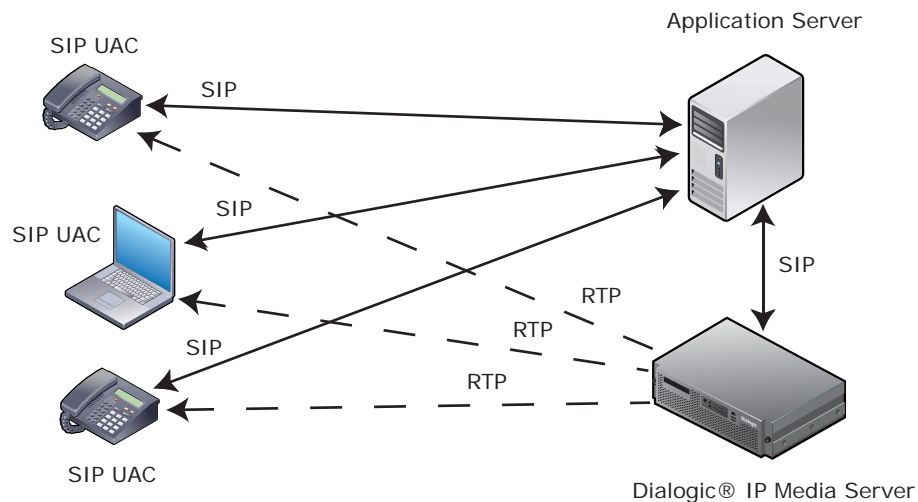
A control agent, such as a softswitch or an application server, can use a call control (signaling) protocol to request a service from the IP Media Server and to set up media streams between the IP Media Server and SIP-enabled endpoint devices. SIP (Session Initiation Protocol) is the signaling protocol currently used for call control on the IP Media Server.

In the course of signaling, the application server can perform the following:

- ◆ Request the service
- ◆ Set up the connection
- ◆ Modify the connection, if necessary
- ◆ Tear down the connection

If such a connection is established, the IP Media Server is directly engaged with the endpoint devices in the media plane through third party call control. The IP Media Server, however, is not visible to the endpoint devices in the signaling plane.

Figure 1 illustrates the call control relationship.



**Figure 1. Call Control Architecture**

The external application specifies what constitutes a call, not the IP Media Server. Upon receiving a request, the IP Media Server call control engages the correct application for processing.

The IP Media Server is controlled by the following major control protocols and scripting languages:

- ◆ [Session Initiation Protocol \(SIP\)](#)
- ◆ [Session Description Protocol \(SDP\)](#)

- ◆ [Media Server Control Markup Language \(MSCML\)](#)
- ◆ [VoiceXML](#)

## Session Initiation Protocol (SIP)

SIP is a peer-to-peer signaling protocol that employs a request and response model to create, manage, and tear down single or multiple concurrent sessions for the delivery of services.

### Session

A session is defined as having at least two endpoints with one or more bi-directional IP streams to, from, or within the IP Media Server.

A control agent can initiate and control the following services with SIP messages:

- ◆ [Network Announcements](#)
- ◆ [Conferences](#)
- ◆ [IVR](#)
- ◆ [VoiceXML](#)

### Service Indicators

The IP Media Server employs a service indicator in the user address portion of the Request URI. This indicator specifies the service to which a request is being addressed and can be one of the values listed in Table 1.

Table 1. SIP Service Indicators

Service Indicator	Definition
annc	Announcements
conf	Conferencing (simple and advanced with MSCML)
ivr	IVR with MSCML
dialog	VoiceXML

If the user address portion of the service indicator is not one of the supported service indicators, the default service, `dialog`, is invoked. You can configure the default application on the SIP page of the Web User Interface (Web UI).

The IP Media Server takes advantage of the inherent flexibility provided by the SIP URI conventions and by SIP's acceptance of message-body payloads. For example, when SIP is extended with the MSCML scripting language, SIP requests enable control of advanced conferences as well as Interactive Voice Response (IVR) through the `ivr` service.

For details about SIP, see [Chapter 2, "Session Initiation Protocol \(SIP\)"](#).

## Session Description Protocol (SDP)

SIP uses the Session Description Protocol (SDP) to convey information about media streams. The IP Media Server supports parsing and formatting for SDP mandatory elements, as well as several optional elements.

For more details about SDP, see [“Session Description Protocol” \(page 79\)](#).

## Media Server Control Markup Language (MSCML)

Media Server Control Markup Language (MSCML) is a Dialogic-developed markup language used to extend SIP requests for IVR and advanced conferencing functions.

There are six types of MSCML requests which are grouped into two basic categories:

- ◆ Requests that support advanced conferencing (conf service)
- ◆ Requests that support interactive voice response (ivr service)

Table 2 lists both the MSCML conferencing requests and the MSCML IVR requests, and also shows the pages where these requests are defined.

Table 2. MSCML Conferencing and MSCML IVR Requests

MSCML Conferencing	MSCML IVR Requests
<a href="#">configure_conference (page 134)</a>	<a href="#">play (page 165)</a>
<a href="#">configure_leg (page 136)</a>	<a href="#">playcollect (page 165)</a>
	<a href="#">playrecord (page 166)</a>
	<a href="#">stop (page 166)</a>

For details about these MSCML requests and services, see [Chapter 4, “Conferencing API”](#).

## SIP Methods with MSCML

The SIP INVITE method is used to convey the desired session parameters from the application server to the IP Media Server.

For mid-call changes, the XML payload is sent using SIP INFO.

---

**Note:** To provide that XML payloads are delivered in the correct order, the application server must support sequenced INFO messages for MSCML (see [“CSeq” \(page 71\)](#)).

---



## Payload

The size of the MSCML payload is relatively small and is appropriate for inclusion in the SIP body. Each MSCML payload contains a single request or response, and each SIP `INVITE` or `INFO` carries, at most, one MSCML body part.

Most MSCML request attributes have default values or are defined as `#IMPLIED`. MSCML attributes can be omitted from the request if they are not needed.

For further details, see [“MSCML Schema” \(page 367\)](#).

The Multi-Internet Message Extension (MIME) type used to describe MSCML content is `application/mediaservercontrol+xml`.

## Responses

MSCML responses are carried back to the application server in the body of the SIP request message. Responses to MSCML requests are also defined in the Document Type Description (DTD) using a simple form of XML. The response is easy to process and does not require a full XML parser on the application server.

## Events

Applications subscribe to `activetalker` and asynchronous Dual Tone Multi-Frequency (DTMF) event notifications through MSCML directives. Events are reported within the SIP session that subscribed to the event.

---

Note: For details on creating advanced conferences with MSCML, see [Chapter 4, “Conferencing API”](#) and [Chapter 5, “IVR with MSCML”](#).

---

## VoiceXML

Voice Extensible Markup Language (VoiceXML) is designed specifically for speech-based telephony and video applications. VoiceXML audio dialog scripts can include a mix of digitized audio clips and DTMF inputs.

A SIP `INVITE` request directed to the dialog service directs the VoiceXML interpreter to retrieve a URI-specified VoiceXML script and translate the script-directed commands.

The IP Media Server supports VoiceXML 1.0 and VoiceXML 2.0.

The following table presents the IP Media Server functionality supported by VoiceXML 1.0 and VoiceXML 2.0.

Table 3. Functionality Supported by VoiceXML 1.0 and VoiceXML 2.0

Functionality	VoiceXML 1.0	VoiceXML 2.0
MRCP 1.0 for Automated Speech Recognition (ASR) support	No	Yes
MRCP 1.0 for Text to Speech (TTS) support	No	Yes
RTSP	No	Yes
T.30 fax detection	Yes	Yes
T.38 fax detection, termination, and initiation	Yes	Yes
Text overlay	No	Yes

For further details about VoiceXML 1.0, see [“VoiceXML Version 1.0 and Dialog Service”](#) (page 319). For further details about VoiceXML 2.0, see [Chapter 6, “VoiceXML Version 2.0 and Dialog Service”](#).

# Media Storage, Processing, and Supported Codecs

The IP Media Server supports several audio and video codecs, and stores and retrieves content in several encodings and file formats. Supported audio codecs are summarized in [Table 4](#).

Table 4. Supported Audio Codecs

Codec	RTP payload type	Mode	Bandwidth (Kbps)	Packet time (ms)
AMR-NB	96... 127	0	4.75	20, 40
		1	5.15	
		2	5.9	
		3	6.7	
		4	7.4	
		5	7.95	
		6	10.2	
		7	12.2	
G.711 ulaw	0	N/A	64	10, 20, 30
G.711 alaw	8	N/A	64	10, 20, 30
G.726	2	N/A	32	10, 20, 30
G.729	18	N/A	8	10, 20, 30

Supported video codecs are summarized in [Table 5](#).

Table 5. Supported Video Codecs

Codec	RTP Payload Type	Profile and Level	Maximum bit rate (Kbps)	Maximum frame rate (fps)	Frame size
H.263	34	Baseline (0) Level 10, 20, 30	384	30	CIF, QCIF
H.263-1998 (H.263+)	96...127	Baseline (0) Level 10, 20, 30	384	30	CIF, QCIF
H.263-2000 (H.263++)	96...127	Baseline (0) Level 10, 20, 30, 45	384	30	CIF, QCIF
H.264	96...127	Baseline Level 1, 1b, 1.1, 1.2, 1.3	768	30	CIF, QCIF

## Audio

For all services, the IP Media Server can receive and transmit RTP voice packets encoded as G.711 $\mu$ -law, G.711a-law, G.726, G.729AB, and AMR-NB. See [Table 4 \(page 35\)](#).

You can license the G.726, G.729AB, and AMR-NB codecs to run on the host.

**G.729** The IP Media Server supports G.729AB as an RTP codec. (G.729A for encode/decode. G.729AB for decode only.)

The IP Media Server automatically converts (transcodes) stored content accessed via file:/// or http:// URLs from the content encoding to the RTP encoding. Any of the available content encodings can be played or recorded using any of the available RTP codecs.

**G.726** The IP Media Server supports G.726 at 32 Kbps as an RTP codec.

**AMR-NB** The IP Media Server supports AMR-NB as an RTP codec. AMR-NB (Adaptive Multi-Rate-Narrow Band) is a multi-mode codec that supports 8 narrow-band speech encoding modes, with bit rates between 4.75 and 12.2 Kbps.

The host-based AMR-NB codec supports true adaptive rates (also referred to as Codec Mode Request, or CMR), so you can specify the mode-set and the far end can select from that list.

## Audio Content Storage

The IP Media Server can retrieve and play audio files with content encoding and file storage formats shown in [Table 6](#).

Table 6. Audio Encoding and File Storage Formats

Content Encoding	File Storage Formats
G.711	raw, .wav, .au
MSGSM	raw, .wav
3GPP (AMR-NB)	.3gp
3GPP2 (AMR-NB)	.3g2

Raw (headerless) G.711 a-law/ $\mu$ -law content is represented as \*.alaw and \*.ulaw. Raw (headerless) MSGSM content is represented as \*.msgsm or \*.ms\_gsm.

---

**Note:** If the file format is unknown or unspecified, the Dialogic® IP Media Server assumes headerless  $\mu$ -law.

---

## Video

The IP Media Server can transmit and receive RTP video packets encoded in H.263, H.263+, and H.264. See [Table 5 \(page 35\)](#).

Video is currently supported by the MSCML (conf) and VoiceXML (dialog) services.

Video transcoding is only supported by VoiceXML, and only during playback; it is not supported for recording. Video recordings are done natively. For more information about video transcoding, see [“Video Transcoding” \(page 45\)](#).

## Mixed Audio/Video

---

**Note:** The IP Media Server will not play a video file that is not accompanied by audio. If a file has video that extends beyond the end of the audio track, the playback will be terminated at the end of the audio track.

---

## WAV-based

Mixed audio and video content is supported using a proprietary WAV-based format identified by the MIME type `video/x-wav`. Files of this type can contain a G.711 audio track and an H.263, H.263+, H.263++, or H.264 video track.

This content type is currently supported by the MSCML (conf) and VoiceXML (dialog) service.

## 3GPP and 3GPP2

The IP Media Server supports playback of an associated video and audio stream of 3rd Generation Partnership Project (3GPP) and 3rd Generation Partnership Project 2 (3GPP2) file formats, provided that the video stream is stored in an H.263+, H.263++ or H.264 video track, and the audio stream is stored in a separate AMR-NB audio track.

On playback, the IP Media Server packetizes the video track into RTP streams, and the audio track into RTP packets in one of the supported RTP encodings (G.711, G.726, G.729, AMR-NB).

The IP Media Server also supports recording in the 3GPP file format (it does not currently support recording in 3GPP2). For an example of how to do this in VoiceXML 2.0, see [“MIME Types Supported” \(page 228\)](#).

The MIME type for 3GPP is `video/3gpp`. The MIME type for 3GPP2 is `video/3gpp2`.

The 3GPP and 3GPP2 file formats are currently supported by the MSCML (conf) and VoiceXML (dialog) services.

## Mixed Audio/Video Content Storage

The IP Media Server can retrieve and play multimedia files with content encoding and file storage formats shown in [Table 7](#).

Table 7. Multimedia Encoding and File Storage Formats

Content Encoding	File Storage Formats	Notes
3GPP	.3gp	Audio: AMR-NB Video: H.263+, H.263++, or H.264
3GPP2	.3g2	Audio: AMR-NB Video: H.263+, H.263++, or H.264
WAV-based	.wav	Audio: G.711 Video: H.263, H.263+, H.263++, or H.264

## File Storage and Retrieval

The IP Media Server can process files using NFS, HTTP, or RTSP.

### Network File System (NFS)

The IP Media Server can retrieve, play, store, and record data files using the Network File System (NFS) protocol, as defined in RFC 1094 and RFC 1813.

### HTTP

The IP Media Server can retrieve and play data files using HTTP/1.0 (RFC 1945) and HTTP/1.1 (RFC 2068) servers.

### Real Time Streaming Protocol (RTSP)

The IP Media Server can retrieve and play files stored on a Real Time Streaming Protocol (RTSP) server, as defined in RFC 2326. The files stored on the RTSP server must have audio and video codecs that are supported by the local IP Media Server file playback; see [Table 4 \(page 35\)](#) and [Table 5 \(page 35\)](#).

## DTMF

The IP Media Server supports the in-band tone detection of the DTMF digit set (0-9, #, \*, A-D), as defined in ITU recommendations Q.23 and Q.24, and according to RFC 2833, "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

For more information about DTMF and its use with MSCML, see [“IVR Operations on Participant Legs”](#) (page 126).

## In-Band Busy Tone Detection and Reporting

The IP Media Server supports in-band busy tone detection and reporting. This feature allows you to monitor outbound calls that can terminate on a PBX.

## Services

The IP Media Server supports the following SIP services:

- ◆ [Network Announcements](#)
- ◆ [Conferences](#)
- ◆ [Interactive Voice Response \(IVR\)](#)
- ◆ [Dialog \(VoiceXML\)](#)

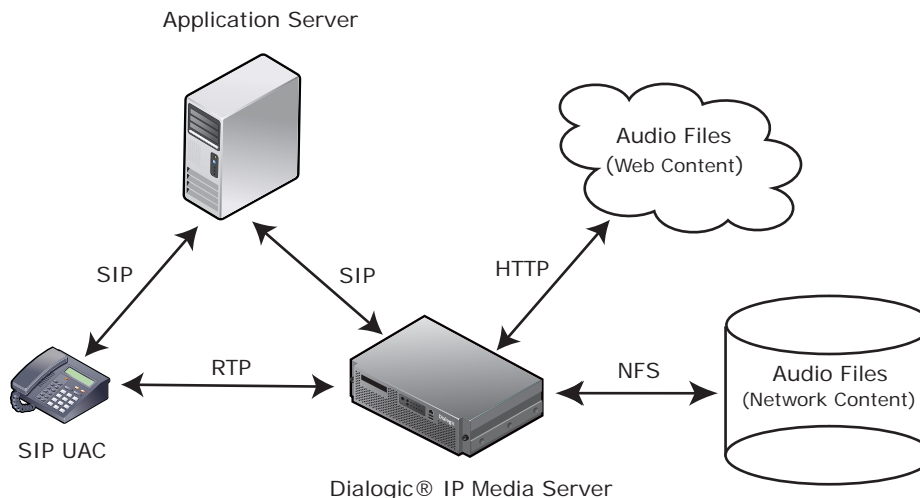
### Network Announcements

Announcement play-out happens when the IP Media Server retrieves stored media and plays it out in an RTP audio stream.

Announcement files are stored at a location accessible by the IP Media Server. The SIP INVITE message that plays an announcement includes the path for the audio file in the `play=` parameter.

Example:

```
INVITE sip:annc@MS_IP;
  play=file:///opt/snowshore/prompts/generic/
  en_us/susan/circuit_busy.ulaw SIP/2.0
```



**Figure 2. Network Announcements**

The IP Media Server is factory-equipped with a library of generic audio segments (see [“Audio Library” \(page 305\)](#)).



## Simple Announcements

Simple announcements are audio files of fixed content that require no user interaction. (For example, a message such as: “All circuits are busy. Please try your call again later.”)

Simple announcements can be stored in a location that is accessible to the IP Media Server and can be in the `file:///` scheme retrieved by NFS or the `http://` scheme retrieved by HTTP.

You can use the Web UI to configure a default location for audio files that play as simple announcements. Once you configure this setting, the IP Media Server prepends it to the `play=` parameter in the SIP message when there is no scheme (`file`, `http`) specified between `play=` and the filename.

For example, if you set your default URL to this:

```
file:///opt/snowshore/prompts/generic/
```

then the IP Media Server interprets this request:

```
INVITE sip:annc@MS_IP;play=circuit_busy.ulaw SIP/2.0
```

as:

```
INVITE
  sip:annc@MS_IP;play=file:///opt/snowshore/prompts/
  generic/circuit_busy.ulaw SIP/2.0
```

## Announcement Sequences and Variable Content Announcements

Announcement sequences are multiple audio files delivered in a single play-out. Each portion of the sequence is a separate file that can be reused in another sequence. In the following example, each sentence can be separate file that can also be used in other sequences:

”Thank you for calling Media Services. The office is currently closed. Our normal business hours are Monday through Saturday, 10AM to 5PM.”

Sequences can also contain variables that are evaluated at run-time. For example, the series of variables

```
<num_dial.wav>, <var1> <changed.wav> <new_num.wav> <var2>
```

becomes

“The number you have dialed, 555-122-2222, has been changed. The new number is 555-122-3333.”

## Conferences

Conferences are categorized in the following way:

- ◆ Simple
- ◆ Advanced

Conferencing requires that two or more RTP streams are mixed so that speakers are heard in the output RTP stream. Input and output channels may be selected from any supported RTP codec. The number of input channels per session ranges from two to the available IP Media Server capacity. The output channels generate a mix representing the inputs for a maximum of three active speakers.

The conference sessions mix the input media streams, creating a set of output streams of mixed media. Advanced conference sessions additionally allow operations on individual inputs and outputs to and from the conference mixer. These operations include:

- ◆ Tone detection
- ◆ Tone generation
- ◆ Tone suppression

### Simple Conferences

The SIP interface supports simple conferencing with no control or special mixing directives. Standard SIP methods create the conference, manage participant access, and delete the conference. Based on IP Media Server capacity, participants can listen and talk, with a maximum of three active talkers.

The number of participants per simple conference is dynamically allocated.

The URI in the SIP `INVITE` contains a conference service indicator (`conf`) with a unique instance identifier. For example:

```
sip:conf=conference1@MS_IP SIP/2.0
```

The conference service creates a conference for the first instance of the unique service instance identifier. The IP Media Server places subsequent requests with the same service instance identifier into that conference.

### Advanced Conferences

Enhanced SIP (inclusion of MSCML), supports the following:

- ◆ Advanced conferencing with call control
- ◆ IVR prompting
- ◆ Mixing directives
- ◆ Event notification

For example, you can set parameters on a per-conference basis to record the conference and play announcements to all participants.

## Interactive Voice Response (IVR)

The IVR service supports interactive voice response applications with functions for playing prompts, collecting DTMF digits, and recording. This service uses Media Server Control Markup Language (MSCML) to enable application control of these functions.

## Dialog (VoiceXML)

The dialog service supports interactive voice response and video applications written in VoiceXML. VoiceXML 1.0 and 2.0 are supported on the IP Media Server.

Refer to [“VoiceXML Version 1.0 and Dialog Service” \(page 319\)](#) for information about VoiceXML 1.0.

Refer to [“VoiceXML Version 2.0 and Dialog Service” \(page 182\)](#) for information about VoiceXML 2.0.

## Advanced Video Features

This section describes the advanced video features supported by the Dialogic® IP Media Server:

- ◆ [Video Transcoding](#)
- ◆ [Text overlay](#)

# Video Transcoding

Video transcoding is the process of converting video media from one video codec type to another (for example, from H.264 to H.263) between two endpoints to suit the requirements of the device at each endpoint. Transcoding involves decoding and encoding each frame of a video stream.

In the process of transcoding, features such as frame rate conversion, bit rate control, image resizing, and text overlay can be applied. These features are not applicable if video transcoding is not being used.

Video transrating adjusts the number of video frames per second between two endpoints to suit the requirements of the device at each endpoint.

Bit rate control limits the video input bit rate to a lesser value at the output while applying the algorithms to control video quality.

Image resizing converts video from one image size to another (for example, from CIF to QCIF) between two endpoints to suit the requirements of the device at each endpoint.

In this document, the term video transcoding or video transcoder encompasses video transcoding, video transrating, bit rate control, and image resizing.

For a list of video codecs supported by the video transcoder, see [Table 5 \(page 35\)](#).

---

**Note:** Use of the video transcoder requires an advanced video license from Dialogic. Each license supports a half-duplex (unidirectional) video transcoding session.

---

## Video Transcoder Settings

The video transcoder supports the following settings, which are configured in the Web UI, Configure SIP page:

### Auto

Automatically enables or disables video transcoding on each video connection according to a set of rules (see [Video Transcoder Automatic Setting Rules](#)). This is the default setting.

### Force

Enables video transcoding for each video connection. This option supports as many video connections as there are video transcoding licenses.

### Turn off

Disables video transcoding for each video connection.

### Video Transcoder Automatic Setting Rules

The automatic (auto) setting for video transcoding allows the IP Media Server to determine when video transcoding is required on a video connection and enable it as needed. This setting results in optimal use of video transcoding resources.

A video transcoding license is consumed and video transcoding is enabled for the automatic setting when any of the following conditions is true:

- ◆ The source codec does not match the target codec.
- ◆ The source image size does not match the target image size.
- ◆ The source frame rate is greater than the target frame rate.
- ◆ The source bit rate is greater than the target bit rate.
- ◆ A text overlay is being applied to the video stream.

The video transcoding license is released and video transcoding is disabled when all of these conditions are false.

### Video Transcoder Restrictions

If a session without transcoding is in progress, and playback begins of a new video file whose header indicates that it is not compatible with the endpoint, the IP Media Server continues streaming the video without transcoding.

## Text overlay

Text overlay provides the ability to superimpose multiple text elements over a video stream. The resulting video stream contains both the text elements and the original video stream.

The text elements can be of varying fonts, sizes and colors; and they can be inserted in real time. For example, you can create an interactive menu or display corporate information as a text overlay over a video stream.

Text overlay functionality is implemented in VoiceXML 2.0.

The IP Media Server supports the following text formats for text overlay:

- ◆ ASCII
- ◆ UTF-8 as defined in IETF RFC 3629
- ◆ GB18030 as defined in the Chinese National Standard GB 18030-2005

For more information on text overlay, see [“Text Overlay Functionality” \(page 236\)](#).

---

**Note:** Use of text overlay requires an advanced video license from Dialogic.

---

## Real Time Streaming Protocol (RTSP)

The IP Media Server supports the Real Time Streaming Protocol (RTSP) as a client, as defined in RFC 2326. RTSP is a network control protocol used by RTSP clients to control streaming media servers.

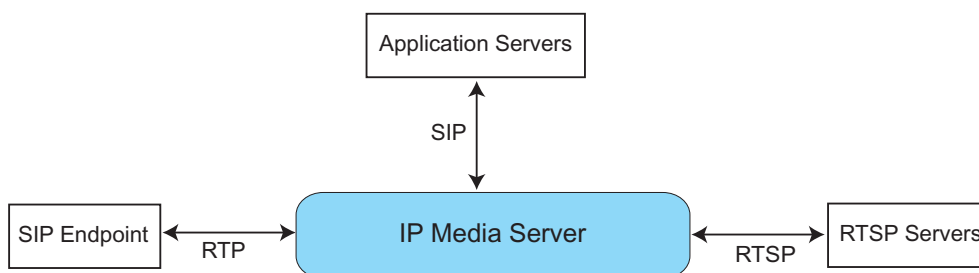
The IP Media Server provides the capability to stream media from RTSP servers to SIP endpoints. This capability is implemented through the RTSP client stack included with the IP Media Server (the RTSP server stack is not included). The SIP endpoints can control playback of the media stream from the RTSP server.

Applications for RTSP include large-scale broadcasts, video voice-mail, and audio/video-on-demand streaming.

RTSP provides a means to choose and establish delivery channels, such as UDP, and to control media sessions between endpoints. RTSP clients can control playback of media using pause, fast forward, and rewind commands. The IP Media Server does not support recording to an RTSP server.

RTSP functionality is supported with VoiceXML 2.0. Properties provided in the VoiceXML 2.0 browser are used to pause, fast forward, and rewind audio and video playback from an RTSP server. For more information, see [“Using VCR and Gain Controls” \(page 300\)](#).

Figure 3 illustrates the role of the IP Media Server in streaming media from an RTSP server to a SIP endpoint.



**Figure 3. Role of the IP Media Server with RTSP Server**

## Configuring Session Keep Alive

The IP Media Server assumes that RTSP servers use RTCP RR (Receiver Reports) for session keep alive.

If an RTSP server uses a different method (listed below) for session keep alive, you will need to specify and configure the following parameters in `snowshore.cfg` for the RTSP client:

### RtspSetParam

Enable or disable SET\_PARAMETER method. Values are: 0=auto, 1=enable, 2=disable. Default value is 2.



### RtspGetParam

Enable or disable GET\_PARAMETER method. Values are: 0=auto, 1=enable, 2=disable. Default value is 2.

### RtspPingFlag

Enable or disable PING flag. This flag controls a ping option used by some servers with GET\_PARAMETER method. Values are: 0=disable, 1=enable. Default value is 0.

### RtspOptionsFlag

Enable or disable OPTIONS flag. Values are: 0=disable, 1=enable. Default value is 0.

The following parameter is used to change the time value for keep alive.

### RtspKeepAliveTimeValue

Time interval in seconds between successive sessions keep alive. Default value is 30 seconds.

## 2 - Session Initiation Protocol (SIP)

This chapter explains in detail the Session Initiation Protocol (SIP) methods, headers, and response codes that the IP Media Server currently supports.

This chapter includes the following major sections:

- ◆ [Conformance](#)
- ◆ [Media Server Availability](#)
- ◆ [SIP Description](#)
- ◆ [SIP Requests](#)
- ◆ [SIP Methods](#)
- ◆ [SIP Headers](#)
- ◆ [SIP Responses](#)
- ◆ [Session Description Protocol](#)
- ◆ [Ports](#)

---

## Conformance

The IP Media Server currently conforms to the following SIP and SDP (Session Description Protocol) specifications:

- [1] IETF RFC 2327, “SDP: Session Description Protocol”, M. Handley et al., April 1998.
- [2] IETF RFC 2543, “SIP: Session Initiation Protocol”, M. Handley, H. Schulzrinne, E. Schooler, J. Rosenberg, March 1999.
- [3] draft-ietf-sip-rfc2543bis-03 “SIP: Session Initiation Protocol”, M. Handley, H Schulzrinne, E. Schooler, J. Rosenberg, March 1999. (Partial conformance for methods, headers, and return codes, as indicated on pages [60](#), [68](#), and [75](#) respectively.)
- [4] IETF RFC 2976, “The SIP INFO Method”, S. Donovan, October 2000.
- [5] draft-ietf-sip-100rel-03, “Reliability of Provisional Responses in SIP”, J. Rosenberg, H. Schulzrinne, March 2001, work in progress.
- [6] draft-ietf-sip-session-timer-04, “SIP Session Timer”, S. Donovan, J. Rosenberg, November 2000.
- [7] draft-ietf-sipping-netann-06, “Basic Network Media Services with SIP”, E. Burger and J. Van Dyke, July 2003, work in progress.
- [8] IETF RFC 2045 and 2046, “Multipurpose Internet Mail Extensions (MIME) Part One: Format of Internet Message Bodies” and “Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types”, Borenstein and Freed, November 1996.
- [9] IETF RFC 5022, “Media Server Control Markup Language (MSCML) and Protocol”, J. Van Dyke, E. Burger, A. Spitzer, September 2007.
- [10] IETF RFC 2833, “RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals”, H. Schulzrinne, S. Petrack, May 2000.
- [11] IETF RFC 3261, “SIP: Session Initiation Protocol”, J. Rosenberg et al., June 2002.

## Media Server Availability

The IP Media Server supports SIP OPTIONS, which allows an endpoint to query the supported methods, content types, extensions, and codecs without establishing a call with the IP Media Server. The IP Media Server extends the SIP OPTIONS response to return IP Media Server specific process availability as well as a snapshot of the current application sessions in use on the IP Media Server. This information is made available when endpoint includes in the SIP OPTIONS request a Content-Type specific only for the IP Media Server.

The IP Media Server supports a specific Content-Type in a SIP OPTIONS request that instructs the IP Media Server to return a set of Media Server Availability information in one XML response from the IP Media Server. This information will be in two Content Types:

- ◆ Licensed Features on the IP Media Server and usage statistics
- ◆ IP Media Server Process Table

Refer to OPTIONS on page 64 for more information on the OPTIONS method and Content-Type (c) on page 71 for more information on this Content-Type.

### Example

The following is an example of a SIP OPTIONS request soliciting the application/media\_server\_usage+xml information from an IP Media Server. The response follows the request below.

### Request

```
OPTIONS sip:192.168.12.160 SIP/2.0
Via: SIP/2.0/UDP 10.128.41.35:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Hammer001 <sip:Hammer001>;tag=9fxced76s1
To: <sip:192.168.12.160>
Call-ID: 3848276298220188511
CSeq: 1 OPTIONS
Accept: application/media_server_usage+xml
Content-Length: 0
```

---

## Response

Note: The SIP OPTIONS response will be sent using UDP and will exceed the 1500 byte limit referenced in RFC 3261 - SIP: Session Initiation Protocol.

---

```
SIP/2.0 200 Ok
Contact: sip:192.168.12.160:5060
Server: SnowShoreMediaServer/0.0.0-0
Via: SIP/2.0/UDP 10.128.41.35:5060;branch=z9hG4bK74bf9
To: <sip:192.168.12.160>
From: Hammer001 <sip:Hammer001>;tag=9fxced76s1
Call-ID: 3848276298220188511
CSeq: 1 OPTIONS
Allow: OPTIONS, INVITE, ACK, INFO, BYE
Accept: application/sdp, multipart/mixed,
        application/mediaservercontrol+xml,
        application/media_server_usage+xml
Supported: timer, 100rel
Content-Type: application/media_server_usage+xml
Content-Length: 3497
```

```
<?xml version="1.0" encoding="iso-8859-1"
?><media_server_usage><features><feature
name="ports" current="0" peak="0" total="750"
/><feature name="mrcp" current="0" peak="0"
total="400" /><feature name="g726" current="0"
peak="0" total="300" /><feature name="g729"
current="0" peak="0" total="300" /><feature
name="h263" current="0" peak="0" total="150"
/><feature name="h264" current="0" peak="0"
total="100" /></features><processes><process
command="fido_start" status="S" parent_pid="18978"
pid="19020"/><process command="vxmlld_start"
status="S" parent_pid="18981" pid="19021"/><process
command="sipd" status="S" parent_pid="18983"
pid="19026"/><process command="logServ" status="S"
parent_pid="18977" pid="19057"/><process
command="monitord" status="S" parent_pid="18984"
pid="19106"/><process command="vxmlld" status="S"
parent_pid="19021" pid="19138"/><process
command="logI" status="S" parent_pid="19021"
pid="19139"/><process command="msinit" status="S"
parent_pid="18980" pid="19160"/><process
command="recoveryd" status="S" parent_pid="18985"
pid="19192"/><process command="uad" status="S"
parent_pid="18982" pid="19216"/><process
command="msprovider" status="S" parent_pid="19160"
pid="19238"/><process command="fido" status="S"
parent_pid="19020" pid="19239"/><process
command="logI" status="S" parent_pid="19020"
pid="19240"/><process command="fido" status="S"
```

```
parent_pid="19239" pid="19241"/><process
command="fido" status="S" parent_pid="19239"
pid="19244"/><process command="fido" status="S"
parent_pid="19239" pid="19245"/><process
command="fido" status="S" parent_pid="19239"
pid="19246"/><process command="fido" status="S"
parent_pid="19239" pid="19248"/><process
command="fido" status="S" parent_pid="19239"
pid="19250"/><process command="fido" status="S"
parent_pid="19239" pid="19252"/><process
command="fido" status="S" parent_pid="19239"
pid="19254"/><process command="fido" status="S"
parent_pid="19239" pid="19255"/><process
command="fido" status="S" parent_pid="19239"
pid="19257"/><process command="fido" status="S"
parent_pid="19239" pid="19263"/><process
command="fido" status="S" parent_pid="19239"
pid="19266"/><process command="fido" status="S"
parent_pid="19239" pid="19267"/><process
command="fido" status="S" parent_pid="19239"
pid="19270"/><process command="fido" status="S"
parent_pid="19239" pid="19271"/><process
command="vxmlld" status="S" parent_pid="19138"
pid="19289"/><process command="vxmlld" status="S"
parent_pid="19138" pid="19290"/><process
command="vxmlld" status="S" parent_pid="19138"
pid="19292"/><process command="mserv" status="S"
parent_pid="19160" pid="20114"/><process
command="dms_start" status="S" parent_pid="18979"
pid="20148"/><process command="dms" status="S"
parent_pid="20148" pid="20168"/><process
command="logI" status="S" parent_pid="20148"
pid="20169"/><process command="ps" status="R"
parent_pid="19238"
pid="20950"/></processes></media_server_usage>
```

# SIP Description

SIP (Session Initiation Protocol) is an IP telephony signaling protocol developed by the IETF. Generally used for voice over IP (VoIP) calls, SIP can also be used for video or any media type; for example, SIP has been used to set up multi-player games.

SIP is a text-based protocol based on HTTP and MIME, which makes it suitable for integrated voice-data applications. Its addressing scheme uses URLs and is human readable; for example: `sip:john.doe@company.com`.

SIP relies on the Session Description Protocol (SDP) for session description and the Real-time Transport Protocol (RTP) for actual transport.

SIP handles the initiation, modification, and teardown of multimedia sessions through the transmission of messages between clients and servers. Messages specify the network path and session description and can also include other payloads, and expected or required options.

SIP allows for the inclusion of parameters and encoded documents in the INVITE message as well as content in the message body. See [“SIP Message Body Types” \(page 57\)](#) for the list of accepted message body types supported in this release.

## SIP User Agents

SIP User Agents are participants in a SIP session.

- ◆ A User-Agent Client (UAC) is a logical entity that issues requests and waits for responses.
- ◆ A User-Agent Server (UAS) is a logical entity that listens for and responds to a SIP request by either accepting or rejecting the request.

UAC and UAS functions change during the course of a call. For example, when playing an announcement, the application server acts as the UAS when it is directly engaged in SIP signaling with UACs such as SIP phones that use media services.

## SIP Message

A SIP message is either a request from a UAC to a UAS identifying a desired service and providing a context for interpreting the request, or a response (status message) from the UAS to the UAC.

Messages employ a text-based protocol, using ISO 10646 character set in UTF-8 encoding. Unless otherwise stated, parameter names and values are case-insensitive.

## SIP Transaction

A SIP transaction consists of the messages from the first request sent by the UAC to the corresponding, final response sent from the UAS. A sequence number (for example, CSeq:2) defines the transactions.

## URLs

Universal Resource Locators (URLs) are used in SIP messages to indicate the originator (From), the routed destination (Request-URI), and final recipient (To) of a SIP request, and also to specify redirection addresses (Contact).

Both dotted IP addresses and fully-qualified domain names are accepted network addresses.

A SIP URL follows the guidelines of RFC 2396 and takes a form similar to a mail to or Telnet URL; for example, user@host.

## Service Indicators

Service indicators in the initial SIP INVITE request direct how the IP Media Server processes the request.

In the context of SIP control, the IP Media Server substitutes a service indicator for the user component on the left-hand side of the URI. For example, a SIP request for an announcement takes the form sip:annc@MS\_IP.

The IP Media Server, upon receiving the INVITE, notes the service indicator and interprets the request accordingly.

Table 8 lists the SIP services and associated indicators.

Table 8. Application Service Indicators

Service	Indicator	Example*	Page
Announcements	annc	INVITE sip: <b>annc</b> @MS_IP; play=(etc.) SIP/2.0	92
Conferencing	conf	INVITE sip: <b>conf</b> =confid@MS_IP SIP/2.0	106
IVR	ivr	INFO sip: <b>ivr</b> @MS_IP SIP/2.0	93
VoiceXML	dialog	INVITE sip: <b>dialog</b> @MS_IP; voicexml=http://path/ filename.vxml SIP/2.0	93 and 183



## Default Application Service

If no service indicator appears in the SIP message, the default application is VoiceXML (the dialog service indicator). You can change this default application through the web interface as follows:

- 1 Select Media Server > Configure > SIP menu
- 2 Set the Default Application under the SIP Parameters section.
- 3 The default version of VXML is 2.0 which you can configure from the Media Server > Configure > VOICEXML menu.

## SIP Message Body Types

Table 9 lists the valid message body content types.

Table 9. Valid Content (MIME) Types

Service / Return Type	Content Type
All services	application/sdp
Advanced conferences (conf= with MSCML) ivr service	application/mediaservercontrol+xml
Advanced conferences (conf= with MSCML)	multipart/mixed
Two pieces to the Return Type: Media Server Licensed Feature Set Media Server Processes and Status	application/media_server_usage+xml

The maximum inbound SIP message size is 2900 bytes. This size prevents buffer overflow caused by proprietary SIP message headers used by some third-party network elements.

## SIP Requests

A SIP Request begins with a request line, followed by one or more headers. An empty line [a carriage return line feed (CRLF) followed by another carriage return line feed] marks the end of the header lines, and can be followed by an optional message body, which can contain a [Session Description Protocol \(SDP\)](#) body.

### Example 1. SIP Message

SIP Request	<pre>INVITE sip:annc@192.168.12.155:5060; play=file:///opt/snowshore/prompts/generic/ circuit_busy.ulaw;early=no SIP/2.0</pre>
SIP Headers	<pre>Via: SIP/2.0/UDP 192.168.1.150:6100 To: &lt;sip:annc@192.168.12.155:5060&gt; From: &lt;sip:test0@192.168.12.153:5060&gt; Call-ID: 27103@192.168.1.150 Contact: sip:192.168.1.150:6100 CSeq: 2 INVITE Content-Type: application/sdp Supported: timer Supported: 100rel Session-Expires: 60 Content-Length: 15</pre>
Message Body	<pre>v=0 o=SnowShoreUaV1 14250 3757 IN IP4 192.168.1.150 s=SnowShore Sdp t=0 0 m=audio 6000/1 RTP/AVP 0 c=IN IP4 192.168.12.154 a=sendrecv a=ptime:10</pre>

## Request Line

The request line consists of the following elements:

- ◆ Method, encoded by ASCII characters, which describes the action requested (INVITE, shown in the above example)
- ◆ Request-URI, which indicates the user or service receiving the request (sip:annc@192.168.12.155:5060, shown in the example)
- ◆ SIP version (SIP/2.0)
- ◆ Parameters and attributes (such as play= and early =), shown in the following example.

### Example 2. SIP Parameters and Attributes

```
INVITE sip:annc@192.168.12.155:5060;  
  play=file:///opt/snowshore/prompts/generic/en_us/  
  susan/circuit_busy.ulaw;  
  early=no SIP/2.0
```

## Headers

A SIP header consists of the header name, followed by a colon, followed by a space and the header value, for example, `Supported: 100rel`.

There are four types of SIP headers:

- ◆ General
- ◆ Request
- ◆ Response
- ◆ Entity

The order of headers is not significant in the SIP Request. See [“Session Description Headers” \(page 79\)](#) for more information on SDP headers.

## Session Descriptor

The session descriptor is encoded in conformance with the Session Description Protocol (RFC 2327) and contains a value or a range of values, such as a media description field and the connection-type attribute.

For details concerning the Session Description Protocol, see [“Session Description Protocol” \(page 79\)](#).

## SIP Methods

The IP Media Server supports five methods from the SIP core protocol, as well as two methods—INFO and PRACK—that have been defined as extensions. Table 10 lists the SIP methods supported by the IP Media Server.

Table 10. Supported SIP Methods

Method	Description	Page
INVITE	Establishes a session or modifies a session.	60
ACK	Acknowledges final responses to INVITE requests.	62
CANCEL	Cancels a pending INVITE that terminates an unestablished call.	63
OPTIONS	Queries the IP Media Server on capabilities and current availability.	64
BYE	Ends a session.	65
INFO	Used for mid-call application requests, responses, and events. (RFC 2976, “SIP INFO Method”)	65
PRACK	Acknowledges receipt of a provisional response. (draft-ietf-sip-100rel-03, “Reliability of Provisional Responses in SIP”)	67

These methods are described in the following sections, including a list of Dialogic-supported headers for each one. (The headers are described in “SIP Headers” (page 68).)

### INVITE

The INVITE method initiates or modifies a session. For example, an INVITE is sent to the IP Media Server requesting that the server participate in a call. An INVITE is followed by an ACK.

SIP INVITEs can also be initiated by the IP Media Server by a VoiceXML script. The IP Media Server typically does not initiate SIP INVITEs, but can act as a refresher for SIP session timers. When the IP Media Server acts as a refresher (user agent client), it sends session timer update messages during a SIP session. (The only other exception is VoiceXML <transfer/>.) For details on the Session Timer, refer to “Basic Network Media Services with SIP” listed in “Conformance” (page 51).

---

**Note:** The IP Media Server supports session timers in compliance with “Basic Network Media Services with SIP”. There is no session timer if early=yes, as a stable session is not established.

---

The INVITE can include the SDP offer. If not present in the INVITE, the IP Media Server provides the SDP in the final response and the client must answer in the ACK.

A re-INVITE renegotiates the media capabilities of an existing session and supports session timers. The re-INVITE must have a higher CSeq than any previous request from the UAC to the IP Media Server.

See the SIP request parameters “[SIP Request Parameters for Announcements](#)” (page 100) for more information.

## Headers in an INVITE

Table 11 lists the mandatory and optional headers in a SIP INVITE method.

Table 11. SIP Headers in an INVITE

	Header	Page
Mandatory	<a href="#">Call-ID (i)</a>	70
	<a href="#">CSeq</a>	71
	<a href="#">From (f)</a>	71
	<a href="#">To (t)</a>	73
	<a href="#">Via (v)</a>	74
	<a href="#">Contact (m)</a>	70
Optional	<a href="#">Accept</a>	69
	<a href="#">Content-Length (l)</a>	70
	<a href="#">Record-Route</a>	72
	<a href="#">Require</a>	72
	<a href="#">Route</a>	72
	<a href="#">Session-Expires</a>	72
	<a href="#">Subject</a>	73
	<a href="#">Supported</a>	73
	<a href="#">Unsupported</a>	74

## ACK

The ACK method acknowledges the receipt of a final response to an INVITE. An ACK functions as follows:

- ◆ Used only with INVITE requests
- ◆ Issued by the party that sent the INVITE
- ◆ Does not generate a response

The ACK request has the same CSeq number as the corresponding INVITE request but comprises a transaction of its own.

The ACK can include SDP information only if the INVITE request did not. A UAC must not send an updated session description in an ACK response, if it has already sent one in the INVITE request.

### Headers in an ACK

Table 12 lists the mandatory and optional headers in a SIP ACK method.

Table 12. SIP Headers in an ACK Method

	Header	Page
Mandatory	Call-ID (i)	70
	CSeq	71
	From (f)	71
	To (t)	73
	Via (v)	74
Optional	Content-Length (l)	70
	Content-Type (c)	71
	Record-Route	72
	Require	72
	Route	72
	Supported	73
	Unsupported	74

## CANCEL

The CANCEL method is a request from the UAC to cancel a pending request with the same Call-ID, To, From, Via header, Request-URI, and CSeq (sequence number only). The IP Media Server ignores a CANCEL request if it arrives after the final response has been sent.

### Headers in CANCEL

Table 13 lists the mandatory and optional headers in a SIP CANCEL method.

Table 13. SIP Headers in CANCEL Method

	Header	Page
Mandatory	<a href="#">Call-ID (i)</a>	<a href="#">70</a>
	<a href="#">CSeq</a>	<a href="#">71</a>
	<a href="#">From (f)</a>	<a href="#">71</a>
	<a href="#">To (t)</a>	<a href="#">73</a>
	<a href="#">Via (v)</a>	<a href="#">74</a>
Optional	<a href="#">Accept</a>	<a href="#">69</a>
	<a href="#">Content-Length (l)</a>	<a href="#">70</a>
	<a href="#">Record-Route</a>	<a href="#">72</a>
	<a href="#">Require</a>	<a href="#">72</a>
	<a href="#">Supported</a>	<a href="#">73</a>
	<a href="#">Unsupported</a>	<a href="#">74</a>

## OPTIONS

The OPTIONS method is a request from the UAC for information from the IP Media Server about the SIP methods supported. The response contains supported methods, content types, and extensions.

### Headers in OPTIONS

Table 14 lists the mandatory and optional headers in a SIP OPTIONS method.

Table 14. OPTIONS Method Headers

	Header	Page
Mandatory	Call-ID (i)	70
	CSeq	71
	From (f)	71
	To (t)	73
	Via (v)	74
Optional	Accept	69
	Content-Length (l)	70
	Record-Route	72
	Require	72
	Supported	73
	Unsupported	74



## BYE

The BYE method terminates a session by indicating that either the application server or the IP Media Server wants to end the connection.

---

Note: BYEs are sent only when a stable session exists.

---

### Headers in BYE

Table 15 lists the mandatory and optional headers in a SIP BYE method.

Table 15. BYE Method Headers

	Header	Page
Mandatory	Call-ID (i)	70
	CSeq	71
	From (f)	71
	To (t)	73
	Via (v)	74
Optional	Accept	69
	Content-Length (l)	70
	Content-Type (c)	71
	Record-Route	72
	Require	72
	Route	72
	Supported	73
	Unsupported	74

## INFO

The INFO method is used during a session to convey application-specific information and events and to communicate mid-session signaling information that does not affect session state.

### Headers in INFO

Table 16 lists the mandatory and optional headers in a SIP INFO method.

Table 16. INFO Method Headers

	Header	Page
Mandatory	Call-ID (i)	70
	CSeq	71
	From (f)	71
	To (t)	73
	Via (v)	74
Optional	Accept	69
	Content-Length (l)	70
	Content-Type (c)	71
	Record-Route	72
	Require	72
	Supported	73
	Unsupported	74

## PRACK

The PRACK method acknowledges receipt of a provisional response. Like ACK, it is used only with INVITE requests and is issued by the party that sent the INVITE. The IP Media Server does not issue this request, but responds with 200 OK to UACs who require or support [5] draft-ietf-sip-100rel-04, “Reliability of Provisional Responses in SIP”.

Figure 4 (page 88) shows a call flow for a provisional response.

### Headers in a PRACK

Table 17 lists the SIP headers in a PRACK Method.

Table 17. Headers in PRACK Method

	Header	Page
Mandatory	Call-ID (i)	70
	CSeq	71
	From (f)	71
	To (t)	73
	Via (v)	74
Optional	Accept	69
	Content-Length (l)	70
	Record-Route	72
	Require	72
	Supported	73
	Unsupported	74

## SIP Headers

There are four types of SIP headers:

- ◆ General headers, used in both requests and responses.
- ◆ Request headers, used for request messages to provide additional information about the request or the UAC.
- ◆ Response headers, which are added to responses to give information that supplements the response code and reason phrase.
- ◆ Entity headers, used to provide additional information about the message body or resource requested.

Table 18 lists the SIP headers that Dialogic supports. The headers are described in “[Supported Headers](#)” (page 69). Unsupported headers are parsed syntactically, but then discarded.

Table 18. Supported SIP Headers

SIP Header	Description	Page
<a href="#">Accept</a>	Indicates acceptable message types.	<a href="#">69</a>
<a href="#">Call-ID (i)</a> (Mandatory)	Identifies a session or a call.	<a href="#">70</a>
<a href="#">Contact (m)</a>	URL where the user can be reached.	<a href="#">70</a>
<a href="#">Content-Length (l)</a>	size of message body.	<a href="#">70</a>
<a href="#">Content-Type (c)</a> (Mandatory)	Media type in the message.	<a href="#">71</a>
<a href="#">CSeq</a> (Mandatory)	Orders different requests within the same session and matches requests against responses.	<a href="#">71</a>
<a href="#">From (f)</a> (Mandatory)	Initiator of the request (name and SIP URL).	<a href="#">71</a>
<a href="#">Max-Forwards</a> (Mandatory)	Upper limit on the number of intermediary proxies.	<a href="#">72</a>
<a href="#">Record-Route</a>	Forces routing through a proxy.	<a href="#">72</a>
<a href="#">Require</a>	Options that must be supported.	<a href="#">72</a>
<a href="#">Route</a>	Forces routing.	<a href="#">72</a>
<a href="#">Session-Expires</a>	Serves as watchdog time out.	<a href="#">72</a>
<a href="#">Server</a>	Provides MS identifying information in IP Media Server request messages.	<a href="#">73</a>

Table 18. Supported SIP Headers

SIP Header	Description	Page
<a href="#">Subject</a>	Indicates client or server.	<a href="#">73</a>
<a href="#">Supported</a>	One or more options.	<a href="#">73</a>
<a href="#">To (t)</a> (Mandatory)	Recipient of the request (name and SIP URL).	<a href="#">73</a>
<a href="#">Unsupported</a> (Mandatory)	Features not supported by the server.	74
<a href="#">User-Agent</a>	Provides IP Media Server identifying information in IP Media Server response messages.	74
<a href="#">Via (v)</a> (Mandatory)	Records the route taken by a request.	74

## Format and Syntax

SIP headers appear after the request line in Requests and after the status line in Responses.

A header line takes the syntax `headername: headervalue`. Allowed values can be a single numeric value, a hex string, or a comma-separated list of values. Values are not case sensitive.

Unsupported headers, malformed headers, and disallowed characters in supported headers do not generate a specific response. They may be ignored, or they may generate a 4xx error (final response indicating client error).

## Supported Headers

This section lists and describes the supported headers in alphabetical order.

Common headers also have a compact form, which appears in parentheses after the header name; for example, `From (f)`, `Contact (m)`.

### Accept

The Accept Header is used in the OPTIONS message to indicate to the Media Server that it would like to request the MS Usage. The format is as follows:

Accept: application/media\_server\_usage+xml

Optional in [BYE](#), [CANCEL](#), [INVITE](#), and [OPTIONS](#) requests.

Optional in response to [BYE](#), [CANCEL](#), [INVITE](#), and [OPTIONS](#) requests.

### Values

Internet media (MIME types of the format type/subtype. See the accepted MIME types in [Table 9 \(page 57\)](#)).

### Call-ID (i)

This general header is created by the UAC to identify a call. It must be globally and chronologically unique across all calls, because it is used to keep track of the SIP session.

The combination of the To, From, and Call-ID headers completely defines a peer-to-peer SIP relationship (referred to in draft-ietf-sip-rfc2543bis-03 “SIP: Session Initiation Protocol” ) as a call leg and in [4b] as a dialog.

The Call-ID is never modified by the IP Media Server.

Mandatory in all requests and all responses.

### Values

A random identifier (case-sensitive string) that is globally and chronologically unique.

A common method of generating Call-IDs is to append the current time and host IP address to a random number.

Example      This identifier is not guaranteed to be unique in the network:  
Call-ID: 123456 .

Example      This identifier is guaranteed to be unique in the network, for example: Call-ID: 20031120105449192.168.15.91@my.com (random number:20031120105; time: 4:49; IP address: 192.168.15.91).

### Contact (m)

This general header contains the URL to which the IP Media Server directs requests in a dialog.

Mandatory for [ACK](#), [INVITE](#), and [OPTIONS](#) requests.

Mandatory in response to [INVITE](#) and [OPTIONS](#) requests.

### Values

All URL parameters are allowed.

### Content-Length (l)

This entity header indicates the size of the message-body sent to the recipient. This header is optional. However, if Content Type is present, the octet count of the message body value must be correct.

Optional for all except **CANCEL** requests (for which it is N/A).

Optional for all responses.

### Values

Decimal number of octets. Any value greater than or equal to 0 is valid.

## Content-Type (c)

This entity header specifies the Internet media (MIME) type in the message body. If an Accept header was listed in the request, the response Content-Type must be listed in the Accept.

Mandatory when a message body is present in the method. Otherwise, optional in all requests and responses and not applicable for **CANCEL** requests.

### Values

Internet media types of the format type/subtype:

- ◆ application/sdp for all services.
- ◆ application/mediaservercontrol+xml for advanced conferences and IVR.
- ◆ multipart/mixed when both application/sdp and application/mediaservercontrol+xml are present.
- ◆ application/mediaserverusage+xml for IP Media Server licensed feature set and IP Media Server processes and status

## CSeq

This general header serves to uniquely identify and order SIP transactions within the same session.

This header contains a 32-bit unsigned integer that increases with every request, except for ACK and CANCEL requests, which use the CSeq number of the INVITE to which they correlate.

A response must contain the CSeq value from the request.

Mandatory in all requests and all responses.

### Values

A decimal number followed by the method name.

## From (f)

This general header indicates the originator of the message and is one of two addresses used to identify the call leg (the other is To: ).

Per draft-ietf-sip-rfc2543bis-03, requests and responses must contain a From general-header field indicating the initiator of the message.

Mandatory in all requests and all responses.

### Values

All URL parameters are allowed.

URL is enclosed in <> when a display name is present.

### Max-Forwards

This request header specifies an upper limit on the number of intermediary proxies.

### Record-Route

This request header forces routing through a proxy. The proxy puts a Record-Route in the request, so that the response is routed back to the proxy.

Optional in all requests and all responses.

### Values

The URL for the required route.

### Require

This request header specifies options the UAC expects the IP Media Server to support. The IP Media Server can also use the Require header in a response.

Optional in all requests and all responses.

### Values

Takes the value of the option(s).

### Route

This request header describes routing for a request.

Optional in all requests and all responses.

### Values

URL parameters for the route.

### Session-Expires

This request header defines a session timer refresh interval.



Optional in [INVITE](#) requests.

Optional in 200 OK responses to [INVITE](#) requests.

### Values

Takes a timer value (in seconds).

### Server

This general header field contains information about the software used by the UAS to handle the request.

### Subject

This general header field provides a summary or indicates the nature of the call and allows call filtering without having to parse the session description. The session description does not have to use the same subject indication as the invitation.

Optional in all responses.

### Supported

This general header enumerates the capabilities (options) of the client or server.

Optional in [BYE](#), [CANCEL](#), [INVITE](#), and [OPTIONS](#) requests.

Optional in response to [BYE](#), [CANCEL](#), [INVITE](#), and [OPTIONS](#) requests.

### Values

Takes the values of the supported options.

### To (t)

This general header indicates the final destination (recipient) of the message.

Per draft-ietf-sip-rfc2543bis-03 “SIP: Session Initiation Protocol”, requests and responses must contain a `To: general-header` field, indicating the desired recipient of the request.

---

**Note:** This header is not used for routing; the Request-URI is used for routing.

---

Mandatory in all requests and all responses.

### Values

All URL parameters are allowed.

URL is enclosed in `<>` when a display name is present.

### **Unsupported**

This general response header indicates features that are not supported by the server.

Optional in all responses.

### **Values**

Takes the values of the unsupported options.

### **User-Agent**

This general header contains information about the UAC originating the request. The purpose of User-Agent is to collect statistics, trace protocol violations, and automatically recognize user agents for tailoring responses to avoid particular user-agent limitations. User-agents should include this field with requests. The field can contain multiple product tokens and comments identifying the agent and any subproducts that form a significant part of the user-agent.

### **Via (v)**

This general header records the route taken by a request. It contains the protocol name and version, type of transport, host name and address, and port number.

Mandatory in all requests and all responses.

### **Values**

All URL parameters are allowed.

URL is enclosed in <> when a display name is present.

# SIP Responses

Responses are returned for most requests. They consist of a status line, several headers, an empty line, and an optional message body.

A response status line consists of three elements:

- ◆ Protocol version
- ◆ Return code
- ◆ Reason phrase (The phrase associated with each code, for example: “SIP/2.0 100 Trying”.)

## SIP Provisional Response Configuration

The IP Media Server can be configured to generate 180 or 183 SIP provisional responses. This feature provides flexibility to address the interoperability requirements of various SIP implementations.

---

**Note:** A provisional response configuration item does not affect the behavior of early media announcements, which by definition, require a 183 to be sent.

---

## SIP Return Codes

All SIP requests other than ACKs are acknowledged with a return code in the response header. Codes are grouped in the following classes:

- ◆ 100-199 indicates a provisional response.
- ◆ 200-299 is a final response indicating success.
- ◆ 400-499 is a final response indicating client error.
- ◆ 500-599 is a final response indicating server error.
- ◆ 600-699 is a final response indicating a global error.

---

**Note:** The IP Media Server does not send redirection responses (class 3xx).

---

Table 19 lists the return codes that can be sent in responses from the IP Media Server, and notes when and why they are sent.

Table 19. SIP Return Codes Generated by the IP Media Server

Code	Code Classes	Reason Phrase	Examples of When / Why
100	Provisional	Trying	INVITE received.
183	Provisional	Session progress	Early media requested in INVITE for announcement (anncc with early=yes).
200	Success	OK	<ul style="list-style-type: none"> <li>◆ Request accepted.</li> <li>◆ CANCEL and BYE received.</li> </ul>
400	Client Error	Bad request	<ul style="list-style-type: none"> <li>◆ Error detected with a message body (such as the failure to execute an XML script).</li> <li>◆ Request received to create a conference that already exists.</li> <li>◆ Request received, but the session is not stable.</li> <li>◆ Request cannot be decoded by the SIP parser.</li> <li>◆ INVITE is improperly formatted. Response is set to the destination indicated by the Via header, if it can be parsed. Otherwise, it is sent to the source IP address:port that generated the request. The Request-URI cannot contain any errors in order for the response to be generated.</li> </ul>
401	Client Error	Unauthorized	Request missing required headers.
403	Client Error	Forbidden	Requested SDP is not supported; for example, G.729a is requested for a conference.
404	Client Error	File not Found	Announcement content not found
405	Client Error	Method not allowed	Stable session established, but UAC attempts to perform an unsupported method.
415	Client Error	Unsupported media type	Requested SDP contains unsupported media type.

Table 19. SIP Return Codes Generated by the IP Media Server (continued)

Code	Code Classes	Reason Phrase	Examples of When / Why
420	Client Error	Bad Extension	INVITE contains Require header specifying an unsupported extension. The only supported extensions are "100rel" (Reliable Provisional Response) and "timer" (SIP Session Timer). The response contains a Supported header that lists these extensions.
480	Client Error	Temporarily unavailable	<ul style="list-style-type: none"> <li>♦ Resource cannot be allocated.</li> <li>♦ Layer of stack not available for processing.</li> <li>♦ The IP Media Server exceeded the number of RTP licenses.</li> </ul>
486	Client Error	Busy Here	<ul style="list-style-type: none"> <li>♦ Contact was successful, but the callee is not willing or able to take the call.</li> <li>♦ Cannot add leg to valid conference.</li> </ul>
487	Client Error	Request terminated	<ul style="list-style-type: none"> <li>♦ Early media requested in an INVITE when announcement has completed. (This is the default response code to end of announcement when early=yes.)</li> <li>♦ Re-INVITE sent while session is being torn down.</li> <li>♦ INVITE sent when CANCEL has been received and processed with a non-stable session.</li> </ul>

Table 19. SIP Return Codes Generated by the IP Media Server (continued)

Code	Code Classes	Reason Phrase	Examples of When / Why
488	Client Error	Not acceptable here	<p>Received SDP information that cannot be processed.</p> <p>The 488 response contains a warning header that describes the details of the SDP negotiation issues, such as an SDP offer containing codecs that are unsupported by the IP Media Server or requests for a media type other than audio.</p>
500	Server Error	Internal error	<ul style="list-style-type: none"> <li>◆ Hardware or software error, or system resources are fully utilized.</li> <li>◆ Internal error processing a request (such as memory allocation or errors from SIP parser API). Maximum concurrent sessions are in use.</li> </ul>
503	Server Error	Service Unavailable	<ul style="list-style-type: none"> <li>◆ System is shutting down.</li> <li>◆ The system has been placed in an operational state in which no new calls are accepted.</li> </ul>
606	Global Error	Not acceptable	<p>The user wants to communicate, but cannot adequately support the session described. Even though the user's agent was contacted successfully, some aspects of the session description, such as requested media, bandwidth, or addressing style were not acceptable. The 606 response might contain a list of reasons in a warning header field, indicating why the session cannot be supported.</p>

# Session Description Protocol

The Session Description Protocol (SDP) provides a text-based format (field=value) for describing how a session is encoded.

SDP supports many types of sessions, from simple audio conversations to multi-media conferences with video and whiteboard support.

SDP content includes session-level information beginning with the v= header line, followed by media-level information, which begins with the m= header.

SDP requires that fields appear in a certain order. Table 20 lists and defines the SDP headers in the required order. Values are case-sensitive.

Examples of uses of SDP in SIP include the following:

- ◆ Version
- ◆ Origin
- ◆ Session
- ◆ Connection
- ◆ Media attributes

Because the type of media session and codec are part of connection negotiation, SIP uses the SDP to specify multiple media types, and can selectively accept or decline each type.

---

**Note:** All SDP must be compliant with ABNF as defined in RFC 2327. Syntax errors are rejected, even when they are present in SDP headers that the IP Media Server ignores.

---

## Session Description Headers

Table 20 describes the SDP description headers and the action taken when they are either received or sent. Headers with a \* next to them are defined as optional in RFC 2327. Actions are described further in Table 23 (page 82).

Table 20. SDP Description Headers

Header	Description	Action (When Received/When Sent)
v	Protocol version which is indicated by 'v=0'.	Ignored/Static Protocol version header is ignored when received in an offer and set to zero on a created offer or response.

Table 20. SDP Description Headers (continued)

Header	Description	Action (When Received/When Sent)
o	Owner/creator and session identifier. o=<username> <session id> <version> <network type> <address type> <address> Typical values for variables:	Ignored/Static The origin header is ignored when received as an offer and set to static and created values.
s	Session name (one per session description)	Ignored/Static Ignored when offered. Static value set to "SnowShore SDP" when created.
*i	Session information	Ignored
*u	URI of description	Ignored
*e	Email address	Ignored
*p	Phone number	Ignored
*c	Connection information. c=<network type> <address type> <connection address> Not required if included in all media descriptions. See "c= (Connection data)" (page 83) for details.	Extracted/Static/Created Extracted when offered. Static when not. Created for late media.
*b	Bandwidth information	Ignored
*z	Time zone adjustments	Ignored
*k	Encryption key	Ignored



Table 20. SDP Description Headers (continued)

Header	Description	Action (When Received/When Sent)
*a	Zero or more attribute lines. See page 84 for details.	Extracted/Static/Created

## Time Description Headers

Table 21 lists and defines the SDP time description headers.

Table 21. SDP Time Description Headers

Header	Description	Action (When Received/When Sent)
t	Time the session is active.	Ignored
*r	Zero or more repeat times.	Ignored

## Media Description Headers

Table 22 lists and defines SDP media description headers.

Table 22. SDP Media Description Headers

Header	Description	Action (When Received/When Sent)
m	Media name and transport address. m=<media> <port> <transport> <fmt list> See “m= (Media information)” (page 82) for details.	Extracted/Static/Created
*l	Media title	Ignored
*c	Connection information. Optional if included at session level. See “c= (Connection data)” (page 83) for details.	Extracted/Static/Created
*b	Bandwidth information	Ignored
*k	Encryption key	Ignored
*a	Zero or more attribute lines. See page 84 for details.	Extracted/Static/Created

## Header Action Classes

Table 23 lists and defines the four different types of SDP header action classes associated with the SDP headers.

Table 23. Action Classes

Header	Definition
Ignored	The header and/or header fields are ignored.
Static	The header and/or header fields are static within the session control stack.
Extracted	The header and/or header fields are extracted from the parsed SIP method or SDP.
Created	The header and/or header fields are dynamic.

## Header Definitions

The following sections describe the SDP headers and their respective attributes.

### **m= (Media information)**

m=<media> <port> <transport> <fmt list>

Attribute	Description
<media> <port>	Media type is set to audio or video when offered, followed by the starting port number. All other media types are ignored.
<transport>	Transport protocol is always set to RTP/AV.

Attribute	Description
<fmt list>	<p data-bbox="618 207 1138 239">Static or dynamic payload codec type(s).</p> <hr/> <p data-bbox="618 285 1411 386"><b>Note:</b> For a list of supported media formats, see “Media Storage, Processing, and Supported Codecs” (page 35) and Table 4 (page 35).</p> <hr/> <p data-bbox="618 432 1414 600">Multiple codecs offered to the IP Media Server will result in a single codec accepted. The choice normally is made from left to right, based on what the IP Media Server can accept. There must be at least one matching codec. Every code can be augmented by RFC 2833 payload type.</p> <p data-bbox="618 617 1414 718">The IP Media Server-offered codecs (one audio, one video) are configurable through the Web User Interface. G.711 <math>\mu</math>-law is the default audio code.</p> <p data-bbox="618 735 1406 867">Use the SIP parameters: Prefer Offer Codec or Require Offer Codec to modify the audio codec selection algorithm. Refer to the Dialogic® IP Media Server <i>Installation and Operations Guide</i> to configure these SIP parameters.</p> <p data-bbox="618 884 1357 984">The payload type for telephone events in the SDP offer is configurable through the Web User Interface. The default payload type is 101.</p>

### c= (Connection data)

c=<network type> <address type> <connection\_address>

Attribute	Description
Network type	IN
Address type	IP4
Connection address	IP address is set to the address designated for the media stream

Connection address setting of 0.0.0.0 is used to specify Hold (suspend audio) on an RTP session. See [“IP Media Server Behavior When Hold Media is Presented” \(page 91\)](#) for additional information on Hold.

---

**Note:** The IP Media Server does not support multicast. A Time-to-Live (TTL) value in the c= field generates an error.

---

## a= (Attribute Lines)

Optional attribute headers enable the inclusion of additional information and can be specified at the session level, media level, or both. A single SDP section can contain multiple attribute fields. The meaning of an attribute is position-dependent:

- ◆ If the a= appears before the first m= field, it is a session-level attribute.
- ◆ If the a= appears after a given m= field, it applies to that media type.

The following attributes are extracted from the SDP body and can also be defined for a session:

Attribute	Description
<b>direction</b>	The direction of a media stream. Possible values include: sendrecv, sendonly, recvonly. Parsed and reported when present and included within formatted SDP with a default value of sendrecv. The offered direction is configurable through the Web UI.
<b>packet time (ptime)</b>	The length of time in milliseconds per packet. Always included with formatted SDP and defaults to 20 ms if not present. The offered ptime is configurable through the Web UI.
<b>dynamic payload type (rtpmap)</b>	Defines a mapping between a codec and a dynamic payload type number. Only supported for RFC 2833 events. All other codecs must use the associated static payload type.

## Ports

### SIP

The IP Media Server listens for incoming SIP messages on the default port, 5060. This port is configurable through the Web User Interface.

### RTP

You can set a starting port number, which is the low value in the range allowed for RTP port negotiation on the IP Media Server. The starting port number is set using the parameter `RtpPortLow` in the `snowshore.cfg` file. `RtpPortLow` can take even-number values from 6000 (the default) to a maximum of 40,000. RTP Ports are allocated on even number ports so you must allow for the addition of two times the number of licensed ports available – not to exceed 40,000. If the parameter is set to a value that falls outside the allowable range, `RtpPortLow` will be set to the default value (6000).

---

**Note:** Changes take effect only after an IP Media Server reset.

---

## Reliability of Provisional Responses

Reliability of Provisional Responses, known as 100 reliability [100rel], is the exchange of acknowledgements for 1xx (greater than 100) class responses (provisional).

The Session Control Stack (SC) asserts 100rel when an initial INVITE method contains the Supported header with the value 100rel, as illustrated in Example 3.

---

**Example 3. Supported Header with 100 Reliability**

```

INVITE
  sip:annc@192.168.1.150:5079;
  play= file:///net/server/hello.wav SIP/2.0
Via: SIP/2.0/UDP 192.168.1.150:5078
To: <sip:annc@192.168.1.150:5079>
From: <sip:abc@192.168.1.150:5078>
Call-ID: 1944@192.168.1.150
Contact: sip:192.168.1.150:5078
CSeq: 18042 INVITE
Content-Type: application/sdp
Supported: timer
Supported: 100rel
Session-Expires: 60
Content-Length: 147

v=0
o=SnowShoreUaV1 12411 30414 IN IP4
  192.168.1.150
s=SnowShore Sdp
t=0 0
m=audio 8000/1 RTP/AVP 0
c=IN IP4 1.1.1.1
a=sendrecv
a=ptime:20

```

The IP Media Server responds with the Require and RSeq headers in any 1xx class method (excluding the 100 Trying method).

```

SIP/2.0 183 Session Progress
Contact: sip:192.168.1.150:5079
Via: SIP/2.0/UDP 192.168.1.150:5078
To:
  <sip:annc@192.168.1.150:5079>;tag=10013564
  03
From: <sip:abc@192.168.1.150:5078>
Call-ID: 1944@192.168.1.150
CSeq: 18042 INVITE
Require: 100rel
RSeq: 32163
Content-Length: 0

```

Once 100rel is negotiated, all subsequent 1xx class methods are acknowledged by the User Agent Client (UAC) with the PRACK method, to which the User Agent Server (UAS) responds with a 200 OK.

Figure 4 shows the call flow with a SIP provisional response. The numbers that follow indicate their respective sections listed inside Figure 4.

- 4 Initial INVITE request contains Supported header with 100rel.
- 5 1xx response contains required 100rel and Rseq headers.
- 6 Each 1xx response requires the UAC to return a PRACK request which makes the Media Server send a 200 OK response.

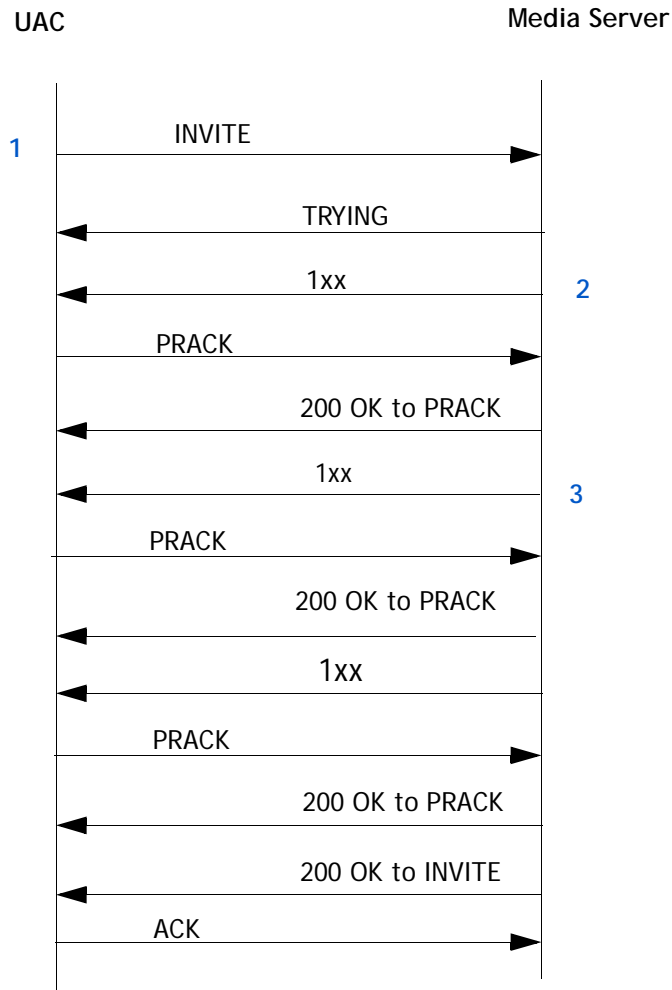


Figure 4. Call Flow: SIP Provisional Response

## Syntax and Escaping

Various sections of the SIP message are permitted to contain different sets of characters. See the SIP specification (RFC 2543) reference for the full description of reserved and unreserved characters allowed in each field. All fields allow the full alphanumeric character set. For convenience, the following tables show which non-alphanumeric characters are considered to be unreserved in the respective fields and headers. Unreserved characters do not need to be escaped.



Table 24 lists the characters that are unreserved in a SIP request, as defined in RFC 2976.

Table 24. Unreserved Characters in SIP Request

Symbol	Name
-	hyphen
_	underscore
.	period
!	exclamation point
~	tilde
*	asterisk
'	single quote
()	parentheses
/	forward slash

Table 25 lists the characters that are unreserved in the user portion (before the @ sign).

Table 25. Unreserved Characters in User Portion

Symbol	Name
&	ampersand
=	equal sign
+	plus sign
\$	dollar sign
,	comma
;	semi-colon
?	question mark
/	forward slash

Table 26 lists the characters that are unreserved for parameters.

Table 26. Unreserved Characters for Parameters

Symbol	Name
[ ]	brackets
/	forward slash
:	colon
&	ampersand
+	plus sign
\$	dollar sign

## Syntax and MSCML Body

In compliance with RFC 1341 (MIME), there must be a blank line (CRLF) following the boundary marker. A CRLF must also separate each line in the multipart/mixed MSCML body. Example 4 represents the correct syntax.

### Example 4. Syntax for MSCML Multi-Part, Mixed Body

```

INVITE sip:conf=123456@10.10.140.1 SIP/2.0
Call-Id: call-12351-1.4.snowshore_conf.5@10.10.221.21
Contact: sip:snow_conf@10.10.221.25
CSeq: 1 INVITE
Expires: 180
From: sip:1.4.snowshore_conf.5@10.10.221.21
Record-Route: <sip:10.10.221.22>
To: sip:conf=123456@10.10.140.1
Via: SIP/2.0/UDP 10.10.221.21:5060
Content-Type: multipart/mixed;boundary=snowbound

--snow-bound
Content-Type: application/mediaserver+xml

<?xml version="1.0"?>
<MediaServerControl version="1.0">
<request>
<configure_conference/>
</request>
</MediaServerControl>

--snow-bound
Content-Type: application/sdp

v=0
    
```

```

o=Pingtel 5 5 IN IP4 10.10.221.21
s=phone-call
c=IN IP4 10.10.221.21
t=0 0
m=audio 8766 RTP/AVP 0 96 8
a=rtpmap:0 pcmu/8000/1
a=rtpmap:96 telephone-event/8000/1
a=rtpmap:8 pcma/8000/1

--snow-bound--

```

## IP Media Server Behavior When Hold Media is Presented

### About Hold SDP

A SIP INVITE/ACK that contains a Hold SDP is a request for the other participant in the call (in this context, the IP Media Server) to cease sending RTP. The IP Media Server recognizes Hold SDP whether it arrives as an SDP offer in the INVITE request or as an SDP response in the ACK request.

Note that, the IP Media Server does not transmit any media streams while on hold.

The IP Media Server remains on hold until a subsequent INVITE/ACK arrives that contains SDP other than Hold SDP.

Hold SDP (as defined in RFC 2543) contains a connection address consisting of (`c=0.0.0.0`).

#### Example 5. Hold SDP

```

v=0
o=SnowShoreUaV1 12853 17864 IN IP4 192.168.12.155
s=SnowShore sdp
t=0 0
c=IN IP4 0.0.0.0
m=audio 4556 RTP/AVP 0
a=sendrecv
a=ptime:20

```

Each participant in the call negotiates Hold independently. The IP Media Server does not automatically answer with Hold SDP when the IP Media Server receives the Hold.

However, when the IP Media Server responds to the initial INVITE to establish a conference Control Leg for an advanced conference, it automatically answers with a HOLD SDP. See [“Advanced Conferencing \(SIP with MSCML\)”](#) (page 92).

For all services, if the initial INVITE/ACK handshake that establishes a session contains Hold SDP, the IP Media Server does not start any media-related activity until a subsequent INVITE/ACK arrives that contains other than Hold SDP.

### Hold Behavior for the Various Media Services

The following sections describe the Hold behavior for various media services.

#### Announcement Service (annc)

After receiving a re-INVITE with Hold SDP, the IP Media Server does not play the announcement immediately. When a subsequent re-INVITE with a valid destination IP address and port number arrives, the IP Media Server plays the announcement.

---

**Note:** This behavior is reliable only when playing content from NFS servers. HTTP retrievals timeout if the Hold duration value exceeds the HTTP's timeout value (fetch value). HOLD is not allowed during early media (`early=yes`) announcement requests.

---

#### Simple Conferencing (SIP)

A participant leg can accept Hold SDP in the initial INVITE/ACK. The session will not be connected to the conference until a subsequent INVITE/ACK is received with other than Hold SDP.

After receiving a re-INVITE with Hold SDP, the IP Media Server removes the session from the conference and stops sending RTP packets.

When the IP Media Server receives a subsequent re-INVITE with a valid destination IP address and port number, RTP is re-established as indicated in a new SDP section and the session is reconnected to the conference.

#### Advanced Conferencing (SIP with MSCML)

The initial INVITE that establishes a conference control leg and creates a conference should contain Hold SDP because the control leg has no associated RTP streams. This INVITE has an MSCML `<configure_conference/>` request in the SIP body. The IP Media Server always returns Hold SDP on the control leg, regardless of whether SDP was sent in the offer.

A participant (or a non-control) leg can accept Hold SDP in the initial INVITE/ACK. The session will not be connected to the conference until a subsequent INVITE/ACK is received with other than Hold SDP.

After receiving a re-INVITE with Hold SDP, the IP Media Server sends a response to any currently executing `<play>`, `<playcollect>` or `<playrecord>` request, indicating `reason=stopped`.

If the IP Media Server receives a subsequent re-INVITE with a valid IP address and port number, the IP Media Server restores the connection to the conference and resumes sending RTP packets to the IP address and port specified.

During Hold, the behavior of any media-related MSCML request received by the IP Media Server is undefined. The `configure_leg` request is independent of the state of the media streams and operates normally while on Hold. The subscription state for asynchronous DTMF and busy events is maintained across Hold/Retrieve transitions.

### **IVR Service (SIP with MSCML)**

If a re-INVITE with Hold SDP is sent, the IP Media Server sends a response to any currently executing `<play>`, `<playcollect>` or `<playrecord>` request, indicating `reason=stopped`.

If the IP Media Server receives a subsequent re-INVITE with a valid IP address and port number, the IP Media Server resumes sending RTP packets to the IP address and port specified.

During Hold, the behavior of any media-related MSCML request received by the IP Media Server is undefined. The `configure_leg` request is independent of the state of the media streams and operates normally while on Hold. The subscription state for asynchronous DTMF and busy event is maintained across Hold/Retrieve transmissions.

### **Dialog Service (VoiceXML)**

For the dialog (VoiceXML) service, the initial script is not retrieved until the IP Media Server receives an INVITE/ACK with non-hold SDP. The startup script is then invoked and VoiceXML processing starts.

The Hold function in a re-INVITE causes the IP Media Server to stop any media-related activity. This function causes the VoiceXML script to block at the next media-related request. The script is paused during Hold and all timers and event processing are disabled.

If the IP Media Server receives a re-INVITE with non-Hold SDP, VoiceXML script execution is restarted at the point where the Hold was received. Any play, record, or collect operation in progress when the Hold was received is also restarted from the point of interruption.

## 3 - Announcement Service API

This chapter describes the differences between simple and sequenced announcements, and outlines the use of variables.

For general information on SIP, see [Chapter 2, “Session Initiation Protocol \(SIP\)”](#). For examples and call flows, see [Chapter 7, “Sample Code and Call Flows”](#).

This chapter includes the following major sections:

- ◆ [Overview](#)
- ◆ [Simple Announcements](#)
- ◆ [Announcement Sequences](#)
- ◆ [SIP Request Parameters for Announcements](#)

---

## Overview

Upon receipt of a SIP request from an application server, softswitch, or proxy server, the Dialogic® IP Media Server uses the Announcement Server API to deliver network announcements.

The Announcement Server API performs the following steps:

- 1 Listens for incoming SIP messages on port 5060. (The default port is configurable through the Web User Interface.)
- 2 Retrieves the files referenced by the `play=` parameter in the SIP request.
- 3 Encodes the retrieved data in an RTP stream.
- 4 Sends the RTP data stream to the destination indicated in the offered SDP.

## File Retrieval

The URI identifies audio files, and these files can be stored anywhere accessible by the Dialogic® IP Media Server, using NFS or HTTP.

## Network File System (NFS)

The Dialogic® IP Media Server stores and retrieves data files on the Network File System server (RFC 1094 and RFC 1813).

Audio files retrieved by NFS must use the `file:///` scheme.

To access Dialogic prompts, the request must reference the following directory on the IP Media Server: `/opt/snowshore/prompts/generic/`.

## HTTP

The Dialogic® IP Media Server retrieves data from HTTP/1.0 (RFC 1945) and HTTP/1.1 (RFC 2068) servers.

Audio files retrieved by HTTP must use the `http://` scheme.

## Announcement Types

The Dialogic® IP Media Server can process two types of announcements:

- ◆ **Simple Announcements**, which are single audio files with fixed content: “Your call did not go through. Please hang up and try again”.
- ◆ **Announcement Sequences**, which are a grouping of audio files identified by a single name. They comprise a series of fixed audio files delivered in a single play-out: “Welcome to Dialogic Corporation. <welcome.wav> How may I direct your call? <direct.wav>”.

Sequences can also contain variables that are evaluated at run-time. For example, “The number you have dialed <num\_dial.wav>, 555-122-2222 <var1>, has been changed. <changed.wav> The new number is <new\_num.wav> 555-122-3333 <var2>”.

## Announcement Service Indicator and Request URI

The announcement service indicator is `annc`.

```
INVITE sip:annc@MS_IP;play=URLname
```

For simple announcements, the Request-URI in the SIP INVITE points directly to the audio file to retrieve and play.

```
INVITE sip:annc@MS_IP;play=file:///opt/snowshore/  
prompts/generic/circuit_busy.ulaw
```

If you have configured a default location for all audio files for simple announcements, the SIP Request-URI will point directly to the file, as shown

```
INVITE sip:annc@MS_IP;play=wrong_number.ulaw
```



## Simple Announcements

Simple announcements are single audio files that have no variables to resolve; for example:

```
INVITE sip:annc@MS_IP;play=http://server/path/  
hangup.wav
```

A simple announcement can be any length. It is termed simple because this type of announcement only requires a basic fetch-and-play operation. For examples and call flows, see [Chapter 7, “Sample Code and Call Flows”](#).

The SIP INVITE that initiates a simple announcement is directed to the announcement service on the IP Media Server, as indicated by the presence of `annc` (the announcement service indicator) in the Request-URI.

There are several [“SIP Request Parameters for Announcements”](#) (page 100). The only required parameter is `play`, which points to the audio file. Other parameters are `early`, `repeat`, `delay`, and `duration`.

## Announcement Sequences

An announcement sequence is a series of audio files played back-to-back without interruption. Each part of a sequence is a unit of recorded information that can be reused in another context.

These units can be a single sound or word (“two”, “hello”), a phrase (“You have selected...”), or one or more sentences (“Our office will be closed today, Friday July 4th. We reopen for business on Monday the 7th.”)

### Variable-Content Announcements

Variable-content announcements are a special type of sequenced announcements that incorporate variables, evaluated at run-time. In addition to listing the audio files to be included in the data stream, the specification for a variable-content announcement references the manner for rendering the variables.

### Implementation

The IP Media Server can process both single audio segments and lists of audio segments which are identified by the MIME type `text/uri-list`. Announcement sequences rely on the IP Media Server's ability to play these arbitrary lists.

The application server (or another application component) provides a Web interface for generating the announcement sequence and returning it to the Dialogic® IP Media Server. The following example shows an application having a small audio segment library which consists of the following files:

File	Contents
welcome.wav	Welcome to...
reached.wav	You have reached...
hours.wav	Our normal business hours are 9AM to 6PM.
name.wav	The Museum of Transportation
closed.wav	The museum is currently closed.
current.wav	We are now offering an exhibit of vintage motorcycles.
website.wav	For a preview of this exhibit, visit our Website: <a href="http://www.transport.com">www.transport.com</a> .

When a call arrives after business hours, the following message plays: “You have reached the Museum of Transportation. The Museum is currently closed. Our normal business hours are 9 AM to 6 PM.” This message can be delivered by the sequence:

reached.wav  
 name.wav  
 closed.wav  
 hours.wav

For explanatory purposes, this portion of the code can be termed the afterhours sequence. The SIP request required to play this announcement would look like:

```
sip:annc@MS_IP;play=http://appserver.carrier.net;prompt=afterhours
```

The IP Media Server first retrieves the sequence of URLs from the application server and then fetches the audio files. The audio file list must include the URL scheme and the full path to the files. For example, if the files are stored on a centralized NFS server in the /var/prompts directory and accessed via NFS, the audio file list would look like this:

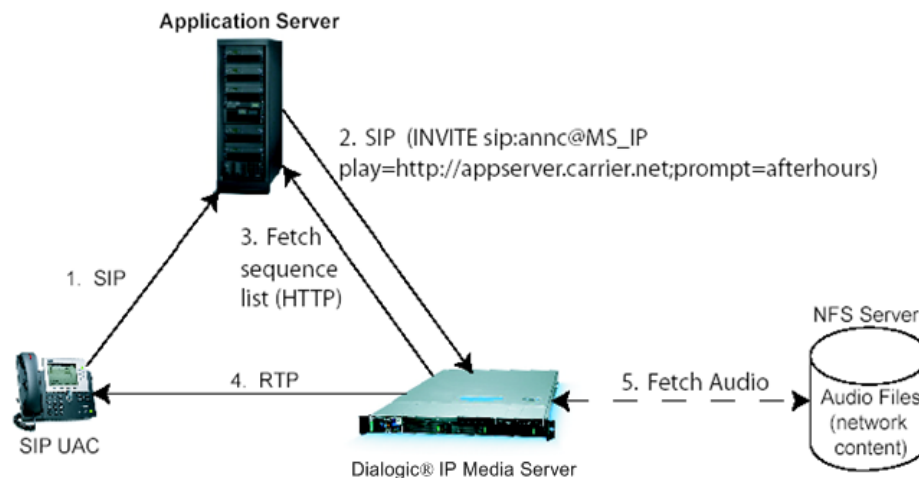
```
file:///net/nfsserver.carrier.net/var/prompts/reached.wav  

file:///net/nfsserver.carrier.net/var/prompts/name.wav  

file:///net/nfsserver.carrier.net/var/prompts/closed.wav  

file:///net/nfsserver.carrier.net/var/prompts/hours.wav
```

The following diagram illustrates an example of a sequenced announcement process. The IP Media Server includes an NFS automounter, so there is no need to explicitly set mount points for prompt directories.



**Figure 5. Playing an Announcement Sequence**

Sequences can also contain variables by including URL(s) to the IP Media Server's onboard phrase server in the returned list. These URLs are of the form:

```
http://localhost/snowshore/phrase.cgi;  

  locale=xxxx&type=yyy&subtype=zzz&value=1234
```

where the URL parameters control what is spoken. For details on supported variables, see [“Variable Types and Subtypes”](#) (page 102).

## SIP Request Parameters for Announcements

The Request-URI contains parameters specifying the audio segment to play and the associated parameter values. Parameters can appear in any order and are separated by semi-colons.

Play is required to reference the audio file. The `early` parameter is optional, but is also the default.

`Repeat`, `delay`, and `duration` are optional parameters and can be used when appropriate.

### play

This parameter specifies the audio resource as a URL. The URL must resolve to a physical audio file or to an HTTP server process, for example servlet that returns a list of audio files using the MIME type `text/uri-list`. This parameter is required. Files are not retrieved unless this parameter is present.

### Values

Value	Description
URI	The complete path for the audio prompt file or servlet, including parameters. Accepts NFS syntax ( <code>file:///</code> ) or HTTP syntax ( <code>http://</code> ).

### Examples

```
play=file:///net/nfsserver.carrier.net/prompts/intro.wav
```

```
play=http://path/audio/allcircuitsbusy.ulaw
```

```
play=http://audioserver.carrier.com;prompt=18
```

**early** This optional parameter specifies whether or not the announcement is to play before the session is completely established.

#### Values

Value	Description
<b>yes</b> <b>(default)</b>	The message is played when the media streams are connected, but before the final response (487) is sent. See <a href="#">“Call Flow for an Early Media Announcement” (page 244)</a> for a call flow example.
<b>no</b>	The announcement plays only after 200 OK.

#### Example

```
early=no
```

#### SIPAnnc404

If the Early parameter above is set to No, you must configure the following parameter in the snowshore.cfg file.

#### Values

Value	Description
<b>yes</b>	For regular announcements, if the announcement file is not found, the IP Media Server will return a SIP 404 Response to the INVITE method.
<b>no</b> <b>(default)</b>	Usual behavior

**repeat** This optional parameter specifies the number of times the announcement or the elements of a sequence should be played in a loop.

#### Values

The allowed range is 1 to 250. Default: 1.

#### Example

```
repeat=2
```

**delay** This optional parameter defines the delay between repeat plays in milliseconds. It has meaning to the IP Media Server only if the repeat parameter is present; otherwise, it is ignored.

### Values

The allowed range is 0 to 30,000 ms in 100 ms increments. Default: 0 [no delay].

### Example

```
delay=20000
```

## duration

This optional parameter defines the maximum duration for the sequence in seconds. Once this amount of time has elapsed, the sequence stops playing. If this is not defined (`duration=0`), the sequence plays to completion.

### Values

The allowed range is 0 to 6000 seconds in increments of 1 second. Default: 0 [no duration limit].

### Example

```
duration=1000
```

## Variable Types and Subtypes

Many standard phrases or items can be referred to using variables. Standard variables are described in this section.

The URI of the desired variable can be specified explicitly in the `play` parameter or can be included as an item in a `text/uri-list` returned by a Web server process.

## date

The value is spoken as a date in the form specified by the subtype. The value is always specified as YYYYMMDD (per ISO 8601, International Date and Time Notation).

**Subtypes**

Value	Description
mdy	20021015 is spoken as “October Fifteenth Two Thousand Two”.
dmy	20021015 is spoken as “Fifteen October Two Thousand Two”.
ymd	20021015 is spoken as “Two Thousand Two October Fifteen”.

**digit**

The value is spoken as a string of digits, one at a time, with one of two phrasings.

**Subtypes**

Value	Description
ndn	is spoken with North American dialing phone number phrasing (NPA-NXX-XXXX), with appropriate pauses.
gen	is spoken as generic digits (one, five, zero).

**duration**

Duration is specified in seconds and is spoken in one or more units of time as appropriate. For example:

Value	Description
3600	is spoken as “one hour”.
3660	is spoken as “one hour and one minute”.
3661	is spoken as “one hour, one minute, and one second”.

**money**

Currency value in USD, spoken in dollars and cents.

**Subtype**

USD (Format: \$\$¢¢)

**Examples**

Value	Description
2500	is spoken as "25 dollars".
25	is spoken as "25 cents".
1	is spoken as "1 cent".
100	is spoken as "one dollar".
101	is spoken as "one dollar and one cent".
1025	is spoken as "ten dollars and twenty five cents".

**month** The specified month in MM format, with 01 denoting January, 02 denoting February, 10 denoting October, and so forth.

**number** A number in cardinal or ordinal form, spoken with one of two phrasings.

**Subtypes**

Value	Description
crd	1 is spoken as "one".
ord	1 is spoken as "first".

**silence** Plays a specified period of silence as indicated by the duration value in milliseconds. For example, the following generates one second of silence:

```
silence:duration=1000
```

**string** Each character of the string is spoken. For example, "a34bc" is spoken as "A, three, four, B, C."

Valid characters are a-z, A-Z, 0-9, #, and \*.

**time** Spoken as a time of day in either twelve or twenty-four hour HHMM format according to ISO 8601, International Data and Time.



**Subtypes**

Value	Description
t12	1700 is spoken as “five p.m.”
t24	1700 is spoken as “seventeen hundred hours”.

**weekday** Spoken as the day of the week. Days are specified as single digits, with 1 denoting Sunday, 2 denoting Monday, and so forth.

## 4 - Conferencing API

This chapter provides information about simple and advanced (enhanced) conferencing.

Simple conferences are controlled and released with standard SIP INVITE and BYE requests. In contrast, advanced conferences are developed by extending SIP with Media Server Control Markup Language (MSCML). This chapter also includes information about MSCML elements and attributes that enable advanced conferencing.

This chapter includes the following sections:

- ◆ [Simple Conferencing](#)
- ◆ [Advanced Conferencing](#)
- ◆ [IVR Operations during a Conference](#)
- ◆ [Simultaneous Play and Record](#)
- ◆ [MSCML Conferencing Requests](#)
- ◆ [Coached Conferencing](#)
- ◆ [MSCML Conferencing Reference](#)

---

Note: If you opt to perform the operations in this chapter, familiarize yourself with the SIP concepts and operations explained in [Chapter 2, "Session Initiation Protocol \(SIP\)"](#).

---

The conference service indicator is conf. An equals sign (=) separates the conference service indicator from a URI or ID that uniquely identifies a conference session, for example: conf=confid@MS\_IP.

---

SIP request URIs identify individual conferences. The URI can be any value compliant with the SIP URI specification, as detailed in RFC 2543. The conference control application must be configured to make sure that the identifier is unique within the scope of any potential conflict.

If the `INVITE` is directed to a URI that does not already exist on the IP Media Server and if the requested resources are available, the IP Media Server creates a new conference.

If the `INVITE` is directed to an existing URI, the IP Media Server interprets the `INVITE` as a request to join the conference represented by that URI.

---

**Note:** The conference URI shared between the application server and IP Media Server can be different from the conference URI visible to participants. This difference provides enhanced security and also simplifies resource management by allowing a specific IP Media Server to be assigned.

---

## Simple Conferencing

The standard SIP Interface (with no MSCML) supports simple conferencing, suitable for calls that do not require the following:

- ◆ In-conference Interactive Voice Response (IVR)
- ◆ Explicit control of mixing mode
- ◆ Event reporting

### How SIP Manages Conferences

Using SIP to manage conferences involves the following methods:

- ◆ The INVITE method to create the conference and join participants to it.
- ◆ The BYE method to remove participants from a conference and delete the conference.

[Table 27 \(page 109\)](#) lists the attributes and default values for simple conferences.

### Creating a Simple Conference

A simple conference is created dynamically if the IP Media Server receives a SIP INVITE request to create a conference using the request URI. For example.

```
INVITE sip:conf=confid@MS_IP
```

If the SIP INVITE lacks an MSCML payload, the IP Media Server sets up a simple conference.

### Adding a Participant to a Simple Conference

To add a participant to a simple conference, direct the INVITE request to the existing conference URI.

While a conference is in progress, participants can leave and rejoin the conference.

The IP Media Server imposes no restrictions on the size of a conference other than the total capacity of the server. If a request to join a conference exceeds the capacity of the IP Media Server, the IP Media Server sends the response: 480 temporarily not available.

### Ending a Simple Conference

A simple conference remains allocated as long as there is at least one SIP session joined to it. When all participants have left the conference, as indicated by the SIP BYE method, the IP Media Server terminates the conference.

## Attributes for Simple Conferences and Participants

Table 26 describes the attributes that are used to create simple conferences for either the entire conference or an individual participant of a conference.

Note: In simple conferences, these attributes take only their default values.

Table 27. Simple Conference Attributes

Attribute	Default Value	Description
type	talker	All participants (legs) are talkers.
mixmode	full	All audio is mixed.
toneclamp	no	Audio is not removed (clamped) when the IP Media Server detects certain repetitive, non-voice signals.
dtmfclamp	no	Detected DTMF digits are not removed from audio for all participants. See the following section for a detailed description concerning dtmfclamp used in a conference leg.

### DTMF Clamping

In a conference leg with DTMF clamping set to `no`, detected 2833 events are regenerated (mixed) to all other conference participants. To receive a 2833 event payload, you must first set up a channel for 2833 mode.

Transmitting 2833 events through conferences is implemented with the concept of a single 2833 talker; only one leg can send 2833 at a time. The leg that first detects 2833 and is not clamped becomes the 2833 talker and has that 2833 event transmitted to completion. Once that event completes, another leg can become the 2833 talker. This scheme prevents mixing of simultaneous 2833 events from different sources thereby confusing the recipients. However, it allows for any leg to send 2833 in the conference, but not simultaneously.

In addition, if a non-2833 leg detects DTMF (and is not clamped), a matching 2833 is sent for each 2833 leg in that conference. However, the reverse is not the case—non-2833 legs will not have DTMF tone inserted on them when 2833 is in the mix.

### Managing Video Switching

The video conferencing service operates as a simple conference which has both audio and video or just audio. The service indicator (`conf`) used for a video conference is the same as for a simple audio conference. An equals sign (=) separates `conf` from a URI or ID that uniquely identifies an instance of a conference session, for example:

```
conf=confid@192.168.12.153
```

where 192.168.12.153 is the IP address or the fully qualified domain name of the IP Media Server.

The only difference in an `INVITE` between joining an audio-only participant to the conference and an audio/video participant is in the SDP section. The audio/video participant's SDP contains a description of both the audio stream and the video stream, for example, `m=video 4028 RTP/AVP 34`.

### Switching Rules

Switching in the conference is based on audio streams. The IP Media Server detects the three loudest talkers from the participants in the conference. The audio mix for the conference is a combination of these three active talkers. The video stream is selected and switched based on these audio active talkers as follows:

- ◆ The first participant to enter a conference is sent video silence (currently an image of a spinning snowflake).
- ◆ After other participants join, the video stream of the loudest talker (with some hysteresis) is sent to all other compatible participants (those using the same video codec). The loudest talker continues to receive the video stream he/she was previously receiving.
- ◆ When a conference is starting up and only one participant has been the loudest talker, every other participant sees the loudest participant. As long as the first participant remains the loudest talker, he/she sees video silence, because there is no previous loudest talker.
- ◆ When a second participant becomes the loudest talker, the first talker then sees that participant, and the new loudest talker sees the previous loudest talker.

### Refresh Frames

Video streams switch only when a special type of frame referred to as a refresh frame ('intra' frame in H.263 and H.263+; 'IDR' frame in H.264) is received by the IP Media Server from the new loudest talker. A refresh frame contains all of the video information required to decode that frame (there is no dependency on data from previous video frames).

### Different Client Types

The IP Media Server supports conferences among clients of different types.

**Conferencing between video-enabled devices and audio-only devices.** Video is never sent to the audio-only device. If an audio-only device becomes the active talker in a conference, each video-enabled device receives the “video silence” stream (currently a looping video clip of a spinning snowflake) in its encoding (H.263, H.263+, H.264).

**Conferencing between H.263, H.263+, and H.264 participants.** Participants only receive a video stream when that stream is of the same encoding type. That is, H.263 endpoints receive only H.263 video streams, H.264 endpoints receive only H.264 video streams, and H.263+ endpoints receive only H.263+ video streams. If the video stream normally sent to an endpoint is not compatible, that endpoint receives only an audio stream.

---

**Note:** For some video clients, when the Media Server stops sending video packets (for example, when an incompatible endpoint becomes the active talker), the video displayed may freeze on the last frame received.

---

## Call Flow and Sample Code Examples

For many types of code samples and call flows for simple conferences, see [“Sample Code and Call Flows”](#) (page 241).

Figure 5 shows an example of the call flow for creating a simple conference (normal media).

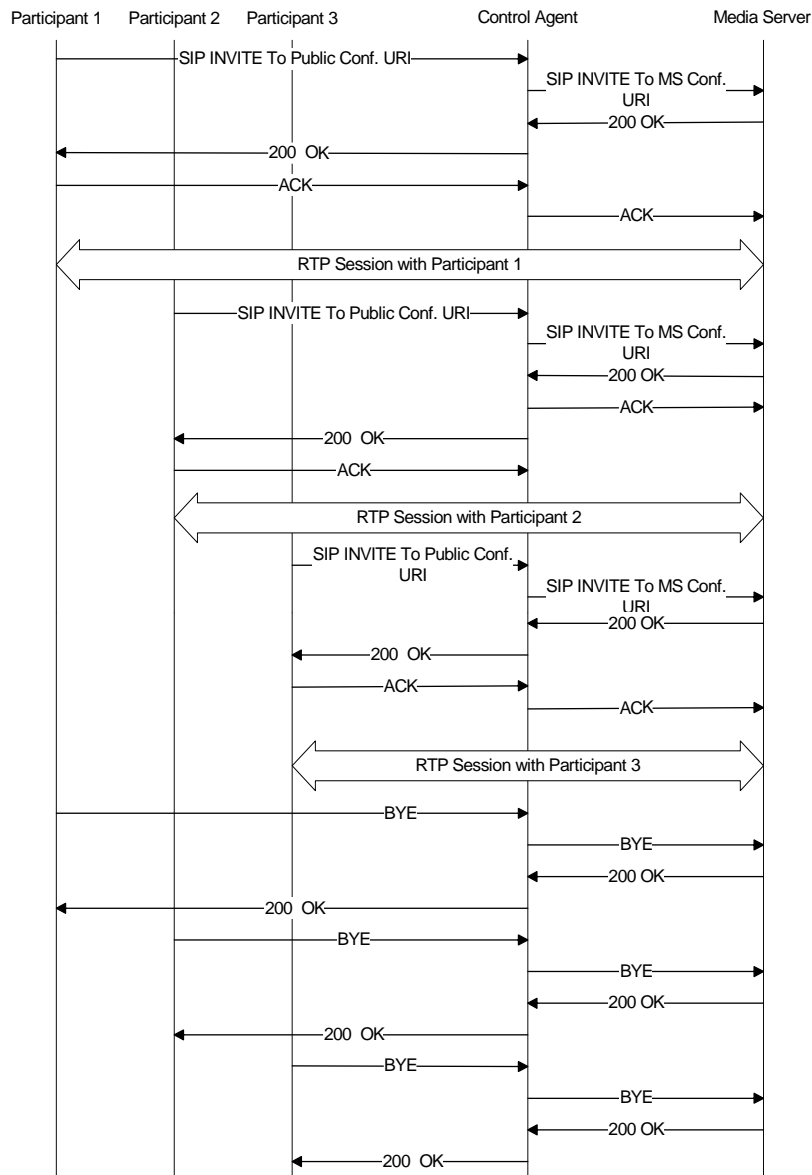


Figure 6. Call Flow for Simple Conference (Normal Media)

## Code for Creating a Simple Conference

Example 6 illustrates a simple conference.

Example 6. Creating a Simple Conference

```
INVITE sip:conf=1234@192.168.12.153 SIP/2.0
From: sip:threepcc@192.168.1.126;tag=1as8ut
To: sip:conf=1234@192.168.12.153
Call-ID: 1031579120515@192.168.1.126
CSeq: 1327320033 INVITE
```



---

```
Content-Length: 201
Content-Type: application/sdp
Contact: <sip:192.168.1.126:5060;
  transport=udp>
Via: SIP/2.0/UDP 192.168.1.126:5060
```

```
v=0
o=Pingtel 5 5 IN IP4 192.168.12.109
s=phone-call
c=IN IP4 192.168.12.109
t=0 0
m=audio 8770 RTP/AVP 0 96 8
a=rtpmap:0 pcmu/8000/1
a=rtpmap:96 telephone-event/8000/1
a=rtpmap:8 pcma/8000/1
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
To: sip:conf=1234@192.168.12.153;tag=1031665657
From: sip:threepcc@192.168.1.126;tag=1as8ut
Call-ID: 1031579120515@192.168.1.126
CSeq: 1327320033 INVITE
Contact: sip:192.168.12.153:5060
Content-Type:application/sdp
Session-Expires:120
Content-Length: 153
a=ptime:20
```

```
v=0
o=SnowShoreUaV1 22263 30720 IN IP4 192.168.12.153
s=SnowShore Sdp
t=0 0
m=audio 4202 RTP/AVP 0
c=IN IP4 192.168.12.154
a=sendrecv
```

## Advanced Conferencing

Extending SIP with MSCML allows developers to create complex conferencing applications.

Advanced (also called enhanced) conferencing allows an application to do the following:

- ◆ Set unique attributes for each participant
- ◆ Play announcements to the conference as a whole
- ◆ Create personalized mixes for each participant
- ◆ Record the conference
- ◆ Subscribe to conference events

### Using MSCML for Advanced Conferencing

Advanced conferences are controlled by MSCML payloads sent in SIP requests. Each SIP INVITE/INFO method contains at most one MSCML body.

The Multipurpose Internet Messaging Extension (MIME) type used to describe the MSCML content is `application/mediaservercontrol+xml`. See [“MSCML Schema” \(page 367\)](#) for the formal definition of the MSCML grammar.

The two major MSCML elements for advanced conference functions are the following:

- ◆ `configure_conference`
- ◆ `configure_leg`

#### **configure\_conference**

When sent in an INVITE message to `conf=confid@MS_IP`, this payload:

- ◆ Creates the conference
- ◆ Determines whether the conference subscribes to Active Talker events (Default: No)

When sent in an INFO message to an existing conference, a `configure_conference` message modifies the properties and event subscriptions for that conference.

#### **configure\_leg**

When sent in an INVITE message to `conf=confid@MS_IP`, this payload joins a participant when the default attributes for the conference are not suitable for that participant.

When sent in an INFO message to the Call ID of a leg, a `configure_leg` message modifies attributes for that participant.

## MSCML Attributes and Elements for <configure\_conference>

The <configure\_conference/> element can control the following elements and attributes:

- ◆ reservedtalkers and reserveconfmedia are set in the initial <configure\_conference> INVITE message and are the only attributes specified in that message.
- ◆ reservedtalkers and subscribe for the entire conference can be modified in subsequent <configure\_conference> INFO requests directed to the URI representing the conference.
- ◆ subscribe element can also be sent in the initial INVITE.

Table 27 describes the attributes and default values for advanced conferences.

Table 28. Advanced Conferencing Attributes

Attribute	Default Value	Description	See Page
reserveconfmedia	yes	Allocates resources to play or record audio to or from the entire conference.	<a href="#">150</a>
reservedtalkers	N/A	Indicates the maximum number of conference participants (legs).	<a href="#">151</a>

Table 28 describes the subscribe element for advanced conferencing.

Table 29. Subscribe Element

Element	Default Description
subscribe	Empty, meaning no event subscriptions are active.

---

Note: Configurable input gain is not currently supported on the IP Media Server.

---

## MSCML Attributes and Elements for <configure\_leg>

Table 29 describes the attributes and their default values for participant legs in an advanced conference. You can modify the following attributes and element for individual participants using the [configure\\_leg](#) INVITE/INFO request.

Table 30. Attributes for Participant Legs in Advanced Conference

Attribute	Default Value	Description	See Page
mixmode	full	Audio from all talker participants is mixed.	<a href="#">148</a>
dtmfclamp	yes	Detected DTMF digits are removed from audio for all participants.	<a href="#">147</a>
toneclamp	yes	Audio is removed (clamped) when the IP Media Server detects certain repetitive, non-voice signals.	<a href="#">151</a>
type	talker	All participants are talker participants.	<a href="#">151</a>

Table 30 describes the subscribe element for participant legs.

Table 31. Subscribe Element for Participant Legs

Element	Default Description
<a href="#">subscribe</a>	Empty, meaning no event subscriptions are active.

## MSCML Attributes for <managecontent>

The <managecontent/> element is used to move recorded content from the IP Media Server to remote locations using the HTTP protocol. This is a store and forward model, which requires the local temporary recording to be completed before it can be sent to the Web server (i.e., <playrecord/>). There is currently no provision for streaming the recording to the Web server while it is being recorded.

Table 31 describes the attributes of the <managecontent/> element.

Table 32. Attributes for <managecontent> in an Advanced Conference

Attribute	Description										
src	Source to be copied, then deleted by the IP Media Server. The URL scheme must be file:///. This attribute is mandatory. It must be in the directory /var/snowshore and must be owned by the user.										
Dest	Destination URL. The URL scheme must be http://. This attribute is not needed when the Action attribute = 'delete'.										
Action	Operation to be carried out. This option can be either 'move' or 'delete'; 'move' is the default. The 'delete' operates on the local source file. After a successful move or delete, the source file is deleted from the IP Media Server. If the request is unsuccessful, the source file is not deleted.										
Httpmethod	HTTP protocol method to be used in request. Only 'post' or 'put' are allowed. The 'post' is the default.										
Name	Field name for the content in the form when using the 'post' method. This is not to be confused with the src or dest attributes. It must be provided when using 'post'. There is no default.										
mimetype	MIME type of content being transferred. If not provided, the IP Media Server tries to infer it based on the following mappings: <table border="1" data-bbox="625 1444 1143 1724"> <thead> <tr> <th>Extension</th> <th>MIME Type</th> </tr> </thead> <tbody> <tr> <td>alaw</td> <td>audio/x-alaw-basic</td> </tr> <tr> <td>.ulaw</td> <td>audio/basic</td> </tr> <tr> <td>.msgsm</td> <td>audio/ms-gsm</td> </tr> <tr> <td>.wav</td> <td>audio/x-wav</td> </tr> </tbody> </table> <p>If the content does not match one of the above, the mimetype attribute must be populated.</p>	Extension	MIME Type	alaw	audio/x-alaw-basic	.ulaw	audio/basic	.msgsm	audio/ms-gsm	.wav	audio/x-wav
Extension	MIME Type										
alaw	audio/x-alaw-basic										
.ulaw	audio/basic										
.msgsm	audio/ms-gsm										
.wav	audio/x-wav										
fetchtimeout	Maximum time permitted for operation in milliseconds. (Default: 10000 ms.)										

## <managecontent> Examples

When the caller is satisfied with the recording, the application commits it to persistent remote storage using <managecontent/>.

- ◆ The application sends:

```

.
.
.
<managecontent
  id="102"
  src="file:///var/snowshore/rec/6A5GH49B.ulaw"
  dest="http://server.carrier.net/recordings/
    myrecording.ulaw"
  action="move"
  method="put"
/>
.
.
.

```

---

**Note:** The application can change the temporary file name assigned by the IP Media Server as part of this operation, as shown.

---

- ◆ To which the IP Media Server responds:

```

.
.
.
<response>
  id="102"
  request="managecontent"
  code="200"
  text="OK"
</response>

```

- ◆ If the request is ambiguous, the IP Media Server returns code="4nn".
- ◆ If the IP Media Server is unable to perform the request, it returns code="5nn". For example, if the file to be deleted either is not in /var/snowshore or is not writable by the IP Media Server, the IP Media Server returns the following:

```

<response>
  id="102"
  request="managecontent"
  code="500"
  text="Bad Request"
  reason=" Delete not permitted."
</response>.

```

- ◆ If there is a network or remote server error, the response shows the remote error within the successful IP Media Server response.

```

<response>
  id="102"

```

```
request="managecontent"  
code="200"  
text="OK"  
<error_info>  
  code="503"  
  text="Service Unavailable"  
  context="http://server.carrier.net/recordings/  
    myrecording.ulaw"  
</error_info>  
</response>
```

## Creating an Advanced Conference

All advanced conferences contain a control participant (leg). The SIP session, which creates the conference, is known as the control leg.

The control leg performs the following:

- ◆ Creates the conference and determines its lifetime. (An advanced conference exists as long as the control leg is present. If the control leg is disconnected, the IP Media Server sends BYEs to all remaining legs, terminating the conference.)
- ◆ Records audio from the entire conference.
- ◆ Plays audio to the entire conference.

If `reserveconfmedia` is set to `yes`, the control leg consumes one leg resource.

The conference control leg is used for conference-wide functions such as active talker reports and playing to or recording from the full conference. It is not used for conference participants, therefore RTP streams are not associated with the conference control leg. Because full conference play and record functionality are internally handled by the media sever, RTP is not involved.

If the application establishes the conference control leg with a SIP INVITE, it must contain "hold" SDP or no SDP. Either of these explicitly states that no RTP streams are associated with the control leg SIP session.

---

**Note:** No RTP is expected on the conference control leg. Therefore, the application should send a hold request (`c=0.0.0.0`), if it prefers sending any SDP. The application does not need to send any SDP.

---

For further details concerning Hold, see [“IP Media Server Behavior When Hold Media is Presented”](#) (page 91).

As with simple conferences, advanced conferences are created dynamically by sending a SIP INVITE to the URI representing the conference. The form is: `sip:conf=confid@MS_IP`.

When the initial SIP INVITE contains an MSCML `configure_conference` payload in the message body, the IP Media Server creates an advanced conference that can be modified with MSCML requests.

If the URI does not already exist on the IP Media Server, but resources are available, the request creates a new conference.

The `configure_conference` message contains the following attributes and elements that control characteristics of the entire conference:

- ◆ `reservedtalkers` (attribute), which sets the maximum number of conference participants.
- ◆ `reserveconfmedia` (attribute), which allocates resources for playing or recording audio to or from the entire conference. (Default: Yes.)
- ◆ `subscribe` (element), which is used if the conference subscribes to `activetalker` events.

The `configure_conference` payload sets the `reserveconfmedia` attribute in the initial INVITE message. This attribute cannot be changed.

To create a conference with 20 reserved talkers, the application specifies the `reservedtalkers` attributes in the initial INVITE.

```
<request>
  <configure_conference reservedtalkers= "20"
    reserveconfmedia="no">
  </configure_conference>
</request>
```

---

**Note:** The `reservedtalkers` attribute is currently ignored by the IP Media Server; so the conference size is not limited by the value of `reservedtalkers`.

---

## Modifying an Advanced Conference

The initial INVITE with a `<configure_conference/>` request defines conference-wide attributes.

The server can change the event subscriptions for the conference. To do this, send a SIP INFO to the URI representing the existing conference. In that SIP request, include a `<configure_conference>` message body with values for the element `subscribe`. For a detailed definition of the `subscribe` element, see page 146.

Example 7 shows a request to modify an `activetalkers` event subscription by changing the default to `yes` with an interval value of 10 seconds.

### Example 7. MSCML Request to Modify Conference Event Subscription

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_conference>
      <subscribe>
        <events>
          <activetalkers report="yes interval=10"/>
        </events>
      </subscribe>
    </configure_conference>
  </request>
</MediaServerControl>
```



```

        </events>
    </subscribe>
</configure_conference>
</request>
</MediaServerControl>

```

Example 8 shows the response to the subscription request.

#### Example 8. Response to Modify Request

```

<?xml version="1.0"?>
  <MediaServerControl version="1.0">
    <response request="configure_conference"
      code="200" text="OK"/>
  </MediaServerControl>

```

## Ending an Advanced Conference

The IP Media Server ends the conference when the control leg receives a BYE method from the controlling application.

Normally, an application server removes all conference participants before deleting the conference.

If one or more participants are still joined to the conference when the control leg receives a BYE, the IP Media Server removes the participants by issuing BYEs on any remaining participant legs to the application server, and then ends the conference.

## Joining Participants (Legs) to an Advanced Conference

To join participants to an existing conference, the application directs a SIP INVITE to that conference URI.

If the default attributes listed in Table 32 are acceptable, the message body can consist of SDP only.

If additional features are required for a participant, the application server sends an INVITE to the conference URI with an MSCML [<configure\\_leg>](#) request that specifies the desired features.

Table 32 describes the attributes and their default values in the [<configure\\_leg>](#) message.

Table 33. Configure\_Leg Attributes

Attribute	Value	Default	Description
dtmfclamp	{yes   no}	Yes	Removes the audio when a DTMF digit is detected.

Table 33. Configure\_Leg Attributes (continued)

Attribute	Value	Default	Description
mixmode	{full   private   preferred   listen   parked}	Full	Specifies the mode for mixing audio from this leg.
type	{talker   listener   internal}	talker	Identifies the participant for this conference leg as a talker.
toneclamp	{yes   no}	Yes	Removes (clamps) audio when the IP Media Server detects non-voice signals.

Example 9 shows a `configure_leg` message body that changes the defaults for `mixmode` and `toneclamp` to `parked` and `no`, respectively.

**Example 9. Joining a Participant with Non-standard Attributes**

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg mixmode="parked" toneclamp="no"/>
  </request>
</MediaServerControl>
```

## Modifying a Conference Participant

Make changes to an existing participant by using a SIP INFO request on the selected call leg that contains a `<configure_leg>` request, indicating the desired attribute. Example 10 shows a `configure_leg` request that mutes the leg.

**Example 10. Modifying a Conference Leg**

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg mixmode="mute"/>
  </request>
</MediaServerControl>
```

## Removing Participants From a Conference

Remove conference participants from a conference when a SIP BYE is received on the corresponding call leg. No MSCML message body is required or expected. The IP Media Server ignores any message body in a SIP BYE request.

## Conference Subsetting

To transfer participants from Conference A to Conference B, while still holding their place in Conference A, create a subset of the existing conference.

To subset an existing conference:

- 1 Create a second conference.
- 2 Put the selected participants on hold.
- 3 Re-INVITE the selected participants to the second conference.

Conference subsetting requires SIP third-party call control operations, but does not require explicit support in MSCML.

To return the participants to the original conference, the application reverses the operation.

## Active Talker Events

The IP Media Server supports active talker events. These are conference-level events requested using the `<activetalkers/>` element, which takes two attributes:

- ◆ `report`
- ◆ `interval`

By default, the `report` attribute is set to `No`, and advanced conferences do not subscribe to active talker events.

To subscribe or turn on active talker events, the application sends a SIP INFO message with `report` set to a `yes` value. For a detailed example, see [“Modifying Conference Using Subscribe” \(page 257\)](#).

With `report` set to `yes`, the active talker event contains the number of talkers in the mix. The event identifies talker participants by their SIP Call IDs, which must be globally unique per SIP [3].

The supported notification method is SIP INFO. Example 11 shows how the XML is generated by the active talker event.

### Example 11. Notification of Active Talker Events

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <notification>
    <conference uniqueID="ab34h76z" numtalkers="3"
      <activetalkers>
        <talker callID="123"/>
        <talker callID="456"/>
        <talker callID="789"/>
      </activetalkers>
    </conference>
  </notification>
</MediaServerControl>
```

## IVR Operations during a Conference

Media Server Control Markup Language supports Interactive Voice Response (IVR) operations such as prompting, DTMF digit collection, and recording. These operations can be used, for example in conjunction with other advanced conferencing features to create complete conferencing applications.

For a detailed description of IVR features and how they are used with MSCML, see [“IVR with MSCML” \(page 153\)](#).

The application can perform IVR operations on either the conference control leg or an individual participant leg. To do this, the application sends INFO requests with the following MSCML commands:

- ◆ `<play/>`
- ◆ `<playcollect/>`
- ◆ `<playrecord/>`
- ◆ `<managecontent/>`

If sending an IVR request within the context of a conference, for example when sending a `<play/>` or `<playcollect/>` message to a participant leg, the application must first send an INFO to that leg, changing its mixmode to parked. After the IVR operation, the application resends an INFO to the leg, changing its mixmode back to full.

In this situation, the application server must wait for the IP Media Server's response to one INFO before sending the next.

---

**Note:** In the above-mentioned scenario, the application server must wait for the IP Media Server's `<playcollect/>` response before sending the `<configure_leg/>` request. Otherwise, the `<playcollect/>` operation may be stopped prematurely.

---



---

**Note:** The MSCML message bodies for play, playcollect, playrecord, and stop can be interrupted. This can occur as a result of an explicit `<stop/>` or if a new MSCML request is sent while the previous one is still executing.

---



---

**Note:** It is not necessary to `park` a conference leg for asynchronous DTMF reporting to function.

---

## Playing and Recording Within the Entire Conference

To play announcements to the entire conference or record the conference output, enable the `reserveconfmedia` attribute and set it to `yes` (default). You set this default value in the `<configure_conference>` request that creates the conference in the initial INVITE.

In every conference:

- ◆ The conference control leg is configured for media handling/recording.
- ◆ If you wish to play media into a conference you are also recording, you can create a separate internal leg for that purpose. For information on how to do this, see [“Simultaneous Play and Record” \(page 133\)](#)
- ◆ Only the conference control leg or a special internal leg can play or record audio in the conference. The conference control leg is counted when calculating the capacity of the IP Media Server.

Set the `mixmode` attribute to `preferred` for this type of playing. Set the `mixmode` to `preferred` to make sure that the audio placed on the conference control leg is always mixed into the conference output.

## Playing to the Conference

The `<play/>` request is typically used to inject audio into a conference; for example, to announce the name of a new participant.

The client application issues this request in a SIP INFO request, using the `<prompt/>` element to specify the audio files to be played.

## Recording the Conference Output

To record, use the `<playrecord>` request. The key attribute of this request is the `recurl`, a URL reference to the target location for the recorded audio. The recording can be preceded as can occur by optional prompt.

There are three timer attributes for the `<playrecord>` request that control when and if a recording is terminated by the absence of speech. These timer attributes are as follows:

- ◆ `initsilence`
- ◆ `endsilence`
- ◆ `duration`

---

**Note:** When recording the output of a conference, these timers generally are disabled, by setting their values to `-1`.

---

You can also use the `<managecontent>` element to move recorded content from the IP Media Server to remote locations using the HTTP protocol.

## Video Conferencing Enhancements

This release provides application developers with the ability to have added greater control over the media mix that is delivered to conference participants. Specifically, a new lecture mode feature has been added which provides that input media from the lecturer is always sent to all participants. Developers may choose whether the lecture properties are applied to the audio input, video input or both.

In addition, it is now possible to mute the audio and video inputs from a participant independently. Both features apply equally to live input or pre-recorded content. These enhancements can be utilized, for example in so-called “video sharing” applications where a pre-recorded video clip is played to the conference participants.

Refer to “[mixmode](#)” (page 148) for the `Configure_Leg` mixmode variations that support these enhancements.

## MSCML Changes

Application developers may now control the output gain on conference legs through MSCML. See the mixmode variants highlighted below.

### **configure\_leg Attribute - Mixmode**

```
<!-- The list of current talkers is used only when
      sending -->
<!-- notifications to the calling application. It
      should never -->
<!-- be set when subscribing. -->
<!ELEMENT configure_leg (inputgain?, outputgain?)>
<!ATTLIST configure_leg
      id CDATA #IMPLIED
      type (talker | listener | internal ) #IMPLIED
      mixmode (full | mute | preferred | parked | private |
      mute_video | mute_audio | lecture | lecture_video |
      lecture_audio) #IMPLIED
      dtmfclamp (yes | no) #IMPLIED
```

## IVR Operations on Participant Legs

Interactive Voice Response (IVR) operations on participant legs include the following requests:

- ◆ `<play/>`
- ◆ `<playrecord/>`
- ◆ `<playcollect/>`

## Detecting DTMF Digits On A Conference Leg

In most cases, a participant leg must be parked before an IVR operation can be executed on that leg. The two exceptions to this are asynchronous DTMF and `<playcollect/>` when no prompt attribute is specified.

### Using Playcollect

Conferencing applications often want to detect DTMF input on a conference leg to trigger some feature, such as dial-out or help. The `<playcollect/>` request collects DTMF digits with or without prompting.

This request causes the IP Media Server to process any digits previously input and possibly wait for additional digits, depending on the attributes supplied. One can use the `cleardigits` attribute to indicate whether the application should remove buffered digits.

Setting the `firstdigittimer` attribute of `<playcollect/>` to `-1` causes the IP Media Server to wait forever for user input. Any buffered digits that were issued prior to the request will be processed.

When digits matching the request specification are detected, a `<response>` is sent to the application. The response contains the collected digits. If the application wants to detect additional DTMF input, another `<playcollect/>` request must be issued.

### Using Asynchronous DTMF

For applications where immediate digit-by-digit notification is desired, the IP Media Server provides asynchronous DTMF event reporting. If using asynchronous DTMF, the application must manage all timers, pattern matching, etc. The asynchronous DTMF feature is used instead of `<playcollect/>`.

---

**Note:** Dialogic does not recommend mixing asynchronous DTMF and `<playcollect/>` for DTMF reporting.

---

To enable asynchronous `keypress` reporting, send the following MSCML payload (shown in Example 12) to the IVR or conference leg on the IP Media Server in a SIP INFO message. This payload is sent for each leg on which digit reporting is desired. Unlike `activetalker` reports, `keypress` reports are not directed to the conference control leg.

#### Example 12. Asynchronous Keypress Reporting

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg>
      <subscribe>
        <events>
          <keypress report="standard|long|both|none"
            maskdigits="yes|no"/>
        </events>
      </subscribe>
    </configure_leg>
  </request>
</MediaServerControl>
```

```

        </events>
    </subscribe>
</configure_leg>
</request>
</MediaServerControl>

```

The report values are the following:

Value	Description
None	Do not report anything (disable reporting if enabled).
Standard	Report digits as they are detected.
Long	Report long digits as they are detected. A long digit is defined as a single key press held down for more than one second or two distinct key presses (a double) of the same digit that occur within two seconds of each other with no other intervening digits.
Both	Report both types of digits. Because a long digit consists of one or more normal digits, a single long duration key press generates one standard event and one long event. A double keypress creates two standard events and one long event. The notification events are sent in a SIP INFO method to the last contact address of record for the session.

The maskdigits values are:

Value	Description
Yes	Disables Clear Text Logging. DTMF data that is normally written into the IP Media Server log files is masked (replaced with asterisks) or discarded.
No (default)	Enables Clear Text Logging.

**Example 13. Keypress Response Attributes and Values**

```

<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <notification>
    <keypress digit="[0-9]|[A-D]|#|*"
      length="standard|long"
      method="standard|long|double"
      interdigittime="{mS}">
      <status command="idle|play|collect|record"
        duration={seconds}/>
    </keypress>
  </notification>
</MediaServerControl>

```



The keypress response (Example 13) contains the following attributes (and the <status> element):

Attribute	Description								
digit	Digit that was detected. Possible values are: 0–9, A–D, #, *								
length	Digit length. Values: standard, long.								
method	Method used to collect digits. Values: <table border="1" data-bbox="602 506 1291 730"> <thead> <tr> <th>Value</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>standard</td> <td>Normal digits</td> </tr> <tr> <td>long</td> <td>Long duration digits (more than 1 second)</td> </tr> <tr> <td>double</td> <td>Double sequence digits</td> </tr> </tbody> </table>	Value	Description	standard	Normal digits	long	Long duration digits (more than 1 second)	double	Double sequence digits
Value	Description								
standard	Normal digits								
long	Long duration digits (more than 1 second)								
double	Double sequence digits								
interdigittime	Time in milliseconds (ms) since the end of the previous digit to the start of this digit (interdigittime). If there is no previous digit, the interdigittime is 0.								

The <status> element has the following attributes:

Attribute	Description
command	Command the IP Media Server was executing at the time the digit was entered. Values: idle, play, collect, record.
duration	How long the IP Media Server has been processing the command (in seconds).

Status information can be used to determine if a digit occurred during a play or a record, or was part of a collect.

## Playing Audio to a Participant Leg

The <play/> and <playcollect/> requests deliver prompts and optionally collect DTMF input on participant legs. These functions normally occur after the user has made a feature request through some DTMF input, as described in the previous section.

To reduce the end-to-end signaling and setup time, the conference interface allows these IVR operations to occur while the user is joined to the conference URI. To turn on this feature, use the mixmode attribute of the <configure\_leg> request.

When set to parked, the mixmode attribute isolates the user's audio input and output so IVR operations can occur. After the IVR operation is completed, the user's mixmode is returned to its previous setting with an additional <configure\_leg> request.

## Detecting and Reporting Busy Call Progress Tones in MSCML

The IP Media Server can detect busy call progress tones and notify the application if they occur. This capability is exposed through MSCML's generic event subscription and notification model. The feature can be used, for example to augment SIP signaling mechanisms when determining if a call has been connected. For example, calls that terminate on a PBX might appear to be connected from a SIP perspective, that is, a 200 final response was generated, but the end device might not have been picked up.

As in the case of other event types, subscriptions and notifications for busy events are transported in SIP INFO messages within an INVITE created dialog. The SIP dialog provides the context and scope for the specific call to which the event subscription pertains.

Busy events are supported by the IVR (*ivr*) and advanced (enhanced) conferencing (*conf*) services. To subscribe to busy events, as shown in the following code sample, send the indicated MSCML payload within the appropriate SIP dialog.

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg>
      <subscribe>
        <events>
          <signal type="busy" report="yes|no"/>
        </events>
      </subscribe>
    </configure_leg>
  </request>
</MediaServerControl>
```

For performance purposes, the IP Media Server executes the busy tone detection algorithm only when the application has subscribed to this event. The application server subscribes to busy events only when they are needed and unsubscribes when they are not. Because this particular signal is useful only during the call setup phase, expect subscriptions to be short, for example, less than 30 seconds.

Busy notifications are sent in the same manner as all other MSCML events. However, the difference between busy events and MSCML events is the MSCML event child element, `<signal/>`. The IP Media Server sends a busy event notification `<notification/>` message after detecting the start of the busy tone.

---

**Note:** Only North American busy tones are currently supported.

---

Example 14 shows the busy event subscription, response and event notification process.

**Example 14. Busy Event Subscription, Response, and Notification Process****Request**

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg>
      <subscribe>
        <events>
          <signal type="busy" report="yes"/>
        </events>
      </subscribe>
    </configure_leg>
  </request>
</MediaServerControl>
```

**Response**

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_leg" code="200"
    text="OK"/>
</MediaServerControl>
```

**Notification**

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <notification>
    <signal type="busy"/>
  </notification>
</MediaServerControl>
```

**Un-subscribe to busy events and response request**

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg>
      <subscribe>
        <events>
          <signal type="busy" report="no"/>
        </events>
      </subscribe>
    </configure_leg>
  </request>
</MediaServerControl>
```

**Response**

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_leg" code="200"
    text="OK"/>
</MediaServerControl>
```

For details concerning the elements, attributes, and values associated with busy event subscription and reporting, see [“MSCML Conferencing Reference” \(page 145\)](#). This reference is an alphabetical listing in tabular format.

---

# Simultaneous Play and Record

You can record a conference while a media stream is being played to it. To do this, use the `configure_leg` method attribute `type` to create an 'internal' leg. That leg can record the conference while another leg is playing to the conference, or vice versa.

## Creating an Internal Conference Leg

To create the internal leg on which to record the conference, send a SIP INVITE request using the MSCML element `configure_leg` with the attribute `type=internal`. When the internal leg is created, you can use it to record the conference.

## Recording a Conference

To record a conference using the internal leg, send an INFO message to the internal leg with the MSCML command `playrecord`, for example:

```
<?xml version='1.0'?>
<MediaServerControl version='1.0'>
  <request>
    <playrecord
      recurl='file:///var/snowshore/rec/temp.ulaw'
      recencoding='ulaw' initsilence="infinite"
      endsilence="infinite"/>
  </request>
</MediaServerControl>
```

The `<playrecord>` MSCML command that initiates the recording must be accompanied by specific command line arguments in order for the recording to take place properly. The `'initsilence="infinite"'` and `'endsilence="infinite"'` attributes must accompany the `<playrecord>` request, so that the recording does not begin or end based on the characteristics of the audio stream. The `'duration="infinite"'` attribute may accompany the `<playrecord>` request, so that the recording does not end before the end of the conference.

The recording ends when either the MSCML `<stop>` request is received or the conference is removed from the IP Media Server, whichever occurs first.

# MSCML Conferencing Requests

This section describes the MSCML request parent and child elements and attributes that support the conferencing service (conf=).

For a complete formal definition of the MSCML grammar, see [“MSCML Schema” \(page 367\)](#).

## Conferencing Request Elements and Attributes

This section describes the MSCML conferencing request elements and their attributes.

### configure\_conference

The <configure\_conference> element can appear in the INVITE or the INFO message.

#### In INVITE Message

The MSCML in the initial INVITE that creates the conference sets the reserveconfmedia and reservedtalkers attributes.

---

**Note:** The reserveconfmedia attribute cannot be changed. Sending the reserveconfmedia in subsequent <configure\_conference> requests to the same conference ID generates a 4xx client error response.

---

A <configure\_conference> request must always be sent to establish an enhanced conference even if the <configure\_conference> request does not have any additional attributes.

---

**Note:** The reservedtalkers attribute is currently ignored by the IP Media Server, so the conference size is not limited by the value of reservedtalkers.

---

Table 33 describes the attributes for an <configure\_conference> element in an INVITE message.

Table 34. Configure\_Conference Attributes in INVITE Message

Attribute	Values	Default	Description
reservedtalkers	2 -available IP Media Server capacity		Specifies the number of legs allocated when the conference is created.

Table 34. Configure\_Conference Attributes in INVITE Message (continued)

Attribute	Values	Default	Description
reserveconfmedia	(yes   no)	Yes	Enables the application to record the conference and to play to the entire conference.  A Yes setting allocates one of the available talker ports as the play leg and sets the mixmode for that leg to preferred. These settings ensure that conference announcements are mixed into the output as the loudest audio.

Table 34 describes the subscribe element.

Table 35. Subscribe Element in INVITE with Configure\_Conference

Element	Description
subscribe	Used when subscribing and unsubscribing to active talker events.

## In INFO Message

A <configure\_conference> element sent in a SIP INFO request to an existing conference can modify the conference size and the subscribe element for the conference. See [Modifying an Advanced Conference on page 120](#) for more information.

Table 35 describes the attributes for an <configure\_conference> element in an INFO message.

Table 36. Configure\_Conference Attributes in INFO Message

Attribute	Values	Description
reservedtalkers	2- available IP Media Server capacity	Specifies the number of legs allocated when the conference is created.

Table 36 describes the subscribe element.

Table 37. Subscribe Element in INFO with Configure\_Conference

Element	Description
subscribe	Used when subscribing and unsubscribing to active talker events.

### configure\_leg

To set the properties for individual conference legs and modify asynchronous DTMF event subscriptions, use the configure\_leg request element. This element can be sent in the following ways:

- ◆ In an INVITE to join a participant whose properties differ from the properties established for the conference as a whole.
- ◆ In an INFO to change the properties for an existing leg.

Table 37 lists and describes the attributes for the configure\_leg element.

Table 38. Configure\_Leg Attributes

Attribute	Value	Default	Description
dtmfclamp	{yes   no}	Yes	Removes the audio when a DTMF digit is detected.
mixmode	{full   private   preferred   listen   parked   mute_video   mute_audio   lecture   lecture_video   lecture_audio}	Full	Specifies the mode for mixing audio from this leg.
type	{talker   listener   internal}	talker	Identifies the participant for this conference leg as a talker.
toneclamp	{yes   no}	Yes	Removes (clamps) audio when the IP Media Server detects non-voice signals.
gain now	delta=db level =db		



Table 38 describes the subscribe element.

**Table 39. Subscribe Element for Configure\_Leg**

<b>Element</b>	<b>Description</b>
subscribe	Used when subscribing and unsubscribing to asynchronous DTMF events.

## Coached Conferencing

The IP Media Server enables applications to create personalized mixes for participants through the MSCML `<configure_team/>` element. A common use of this feature is to support coaching of one participant by another.

The coaching scenario includes three participants:

- ◆ Coach (supervisor) who coaches the agent
- ◆ Agent who interacts with the client
- ◆ Client (customer) who receives advice from the agent.

A coached conference is a 3-way conference call where the coach monitors the call between the company's agent and the external client. The coach hears both the agent and the client speaking; the coach can speak to the agent, but the client cannot hear the coach.

### Overview

To create an application that supports the coaching scenario where the client cannot hear the coach, one would need to identify the relationships among the participants.

You create personalized mixes by manipulating two MSCML objects:

- ◆ The list of team members (`<teammate/>` elements) set using `<configure_team/>`
- ◆ The mixmode attribute set through `<configure_leg/>`

The IP Media Server uses the values of these objects to determine which audio inputs to combine for output to the participant.

In a normal conference, each participant hears the conference mix minus their own input, if they are part of the mixed output. The team list enables the application to specify other participants who can be heard in addition to the normal mixed output. For example, in the coaching scenario, the coach and the agent are configured as part of the team. As a result, the agent can hear the coach as well as the conference mix.

Team relationships are implicitly symmetric. If the application sets the coach as a team member of the agent, then the agent is automatically set as a team member for the coach.

One can use the `id` attribute set through `<configure_leg/>` to identify the various participants. Each participant must have a unique ID to use personalized mixing.

By itself, the team list only defines those participants that can be heard. The `mixmode` attribute of each team member determines whether their audio input is actually included in the personalized mix. If the teammate's `mixmode` is set to `private`, then it is included; if the `mixmode` is set to any other value, it is not. Returning to the coaching scenario, the coach's `mixmode` must be set to `private` to enable the coach's input to be heard only by the agent (student).

## MSCML Elements and Attributes of Coached Conferencing

Coached conferencing requires the <configure\_conference> element and depends on two other major MSCML elements:

- ◆ <configure\_leg> (parent element) with its new value for the mixmode attribute (mixmode=private)
- ◆ <configure\_team> (child element of configure\_leg)

For information about these MSCML elements, see “[configure\\_conference](#)” (page 134) and “[configure\\_leg](#)” (page 136).

### configure\_team

The configure\_team element allows the user to make the participants members of a team within a specific conference.

The configure\_team element is a child of the configure\_leg parent element.

The configure\_team can be sent in either a SIP INVITE message or in an INFO message depending on your application needs.

The configure\_team request can be used in the following situations:

- ◆ In an INVITE message to add a participant whose properties differ from the properties established for the conference as a whole.
- ◆ In an INFO message to change the properties for an existing leg.

Table 39 describes the attributes for the configure\_team element.

Table 40. Configure\_Team Attributes

Attribute	Value	Default	Description
action	add delete query set	query	<p>Allows the user to modify the team list by using one of the following values:</p> <ul style="list-style-type: none"> <li>◆ add—adds a teammate.</li> <li>◆ delete—deletes a teammate.</li> <li>◆ query—returns the teammate list.</li> <li>◆ set—creates a team list when followed by &lt;teammate id= "n"&gt; and also removes all the teammates from the team list, for example, when the creator (originator) of the team list on that specific conference leg wants to remove all of the teammates from the team. If the set operation removes all teammates from a participant, that participant hears the full conference mix.</li> </ul>

Table 40. Configure\_Team Attributes (continued)

Attribute	Value	Default	Description
id	alphanumeric characters; must be less than 64	unique ID (within the conference)	Identifies each participant uniquely in a coached conference.  Note: If the MSCML configure_team specifies an invalid ID, the IP Media Server ignores the invalid ID and turns the action into a query.

## Configuring a Coached Conference

To configure coached conferencing:

- 1 Join each leg to the conference, being certain to include a unique ID in the <configure\_leg/> request.

**Note:** The leg ID needs to be unique only within the scope of the conference it belongs to.

- 2 Configure the teammate list and mixmode of each participant as required.

Table 40 summarizes how MSCML elements and attributes are used to create a coached conferencing configuration and the expected results.

Table 41. MSCML Elements and Attributes for Creating Coached Conferencing

Participant	ID	Team Members	Mixmode	Hears
Supervisor	supervisor	Agent	Private	Customer +Agent
Agent	agent	Supervisor	Full	Customer+ Supervisor
Customer	customer	None	Full	Agent

It is often possible to combine both actions (step 1 and 2) in a single MSCML request.

The following sections describe how to create coached conferencing in a step-by-step approach.

## Creating the Conference

Before joining any participants, the application must create the conference by sending a SIP INVITE which contains an MSCML <configure\_conference/> request with a unique conference identifier.

## Joining and Configuring the Coach

Join the coach leg to the conference and configure its desired properties by sending a SIP INVITE containing a `<configure_leg/>` request. The `<configure_leg/>` element sets the leg's unique ID to `supervisor` and its mixmode to `private`.

The corresponding MSCML request is as follows:

```
<?xml version="1.0"?>
<MediaServerControl version=?1.0">
  <request>
    <configure_leg id="supervisor" mixmode="private"/>
  </request>
</MediaServerControl>
```

The IP Media Server responds as follows:

```
<?xml version="1.0"?>
<MediaServerControl version=?1.0">
  <response request="configure_leg" code="200"
    text="OK"/>
</MediaServerControl>
```

After this step, the conference looks like this:

Participant	ID	Team Members	Mixmode	Hears
Supervisor	supervisor	None	Private	Silence

**Note:** You cannot configure the teammate list for the coach yet because there are no other participants in the conference. A participant must be joined to the conference before it can be added as a teammate for another leg.

## Joining and Configuring the Agent

Join the agent leg to the conference and configure its desired properties by sending a SIP INVITE containing a `<configure_leg/>` request. The `<configure_leg/>` element sets the leg's unique ID to `agent` and sets the supervisor as a team member of the agent. Because team member relationships are symmetric, this action also adds the agent as a team member for the coach.

The corresponding MSCML request is as follows:

```
<?xml version="1.0"?>
<MediaServerControl version=?1.0">
  <request>
    <configure_leg id="agent">
      <configure_team action="set">
        <teammate id="supervisor"/>
      </configure_team>
    </configure_leg>
  </request>
</MediaServerControl>
```

Note: Because the desired mixmode for this leg is full, which is the default value, there is no need to set it explicitly.

---

The IP Media Server responds as follows:

```
<?xml version="1.0"?>
<MediaServerControl version=?1.0">
  <response request="configure_leg" code="200"
    text="OK">
    <team numteam="1">
      <teammate id="supervisor"/>
    </team>
  </response>
</MediaServerControl>
```

After this step, the conference looks like this:

Participant	ID	Team Members	Mixmode	Hears
Supervisor	supervisor	Agent	Private	Agent
Agent	agent	Supervisor	Full	Supervisor

## Joining and Configuring the Client

Join the client leg to the conference and configure its desired properties by sending a SIP INVITE containing a <configure\_leg/> request. The <configure\_leg/> element simply sets the leg's unique ID to customer. No further configuration is required because the desired mixmode, full is the default and the customer has no team members.

The corresponding MSCML request is as follows:

```
<?xml version="1.0"?>
<MediaServerControl version=?1.0">
  <request>
    <configure_leg id="customer"/>
  </request>
</MediaServerControl>
```

Note: Strictly speaking, it is not a requirement that the customer leg be given a unique ID, because it will not be specified as a team member. However, when using coached conferencing, it can be beneficial for each leg to be assigned a unique ID in the initial INVITE request. Assigning a unique ID eliminates the need to set it later by sending a SIP INFO if personalized mixing for the customer leg is desired.

---

After the previous MSCML request is sent, the IP Media Server responds as follows:

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_leg" code="200"
    text="OK"/>
</MediaServerControl>
```

After this last step, the conference configuration is complete and should look like:

Participant	ID	Team Members	Mixmode	Hears
Supervisor	supervisor	Agent	Private	Customer +Agent
Agent	agent	Supervisor	Full	Customer+Supervisor
Customer	customer	None	Full	Agent

## Supervisor Query for Number of Team Members

The IP Media Server responds to a query from the supervisor using numteam to indicate how many members are on the team. Example 15 shows that there are two members on the team, including the requester.

**Example 15.** Shows Two Members on the Team

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_leg" code="200"
    text="OK">
    <team id="super" numteam="1">
      <teammate id="agent"/>
    </team>
  </response>
</MediaServerControl>
```

## Exiting the Conference

If the agent hangs up, the application removes the agent's leg from the supervisor's team list and from the entire conference. This process works exactly like the BYE method—no message is sent to other members of the conference.

For details concerning the BYE method and how a conference is torn down, see [“Ending an Advanced Conference” \(page 121\)](#).

---

**Note:** The IP Media Server does not keep track of old team lists or names of old team lists. Former team members must re-join using the MSCML configure\_team element as initially stated.

---

For sample code and call flows associated with coach conferencing, see [“Coached Conferencing” \(page 270\)](#).

### Using SIP INFO

Coached conferencing can be achieved by sending MSCML `<configure_team/>` elements in SIP INVITE requests. However, if the desired call flow is different or if the initial settings require changes, use SIP INFO to carry the desired MSCML payload.



# MSCML Conferencing Reference

This section describes each of the MSCML conferencing elements and attributes.

## MSCML Elements

This section describes the MSCML conferencing elements (listed in alphabetical order).

---

**Note:** The values for the specific call progress tones are listed under the Type attribute.

---

### activetalkers

Subscribes to activetalker events.

**Parent:** events

---

**Note:** When subscription events are configured for reporting (report=), an updated report is issued only when the activetalkers change, regardless of the interval. If the interval is set to 60 seconds, and the same three participants talk for 119 seconds, there is no new report until the second 60-second interval.

---

### configure\_team

Configures the team list for a conference participant to enable personalized mixes.

---

**Note:** The personalized mix is not activated unless the participant's mixmode is set to private.

---

**Parent:** configure\_leg

### dtmf

Allows the IP Media Server to send DTMF and/or RFC2833 packets provided in the digits attribute.

**Parent:** prompt

**Attributes:**

Attribute	Description
digits	The numbers to play back.
ontime	The amount of time (in milliseconds) that each digit is held.

### Example

```
<?xml version="1.0" encoding="utf-8"?><MediaServerControl  
version="1.0"><request><play><prompt><dtmf digits="123" ontime="200"  
</prompt></play></request> </MediaServerControl>
```

### events

Contains a list of events whose subscription state is to be modified.

**Parent:** subscribe

**Child Elements:** activetalkers, signal, keypress

### keypress

Subscribes to asynchronous DTMF events.

**Parent:** events, notification

### notification

Sent by the IP Media Server to notify the client application when an event is detected.

**Parent:** None

**Child Elements:** signal, keypress, conference

### signal

Used to subscribe for notification of specific call progress tones.

For example, the IP Media Server detects an in-band busy signal and reports this event to the application, so that false connects are eliminated.

---

**Note:** The signal element is also used in the IP Media Server's implementation for Interactive Voice Response (IVR).

---

**Parent:** event, notification

**Values:** busy\*, dial\*, CED, CNG, PVD\_PAMD, PVD, PAMD, interrupt, ring\*, 400

\* North American busy tone

### subscribe

Parent element required for busy event subscription, response, and event notification. To unsubscribe to busy events and response requests, set the signal attribute report to no.

**Parents:** configure\_conference, configure\_leg

**Child Elements:** events

### teammate

Sets up and identifies the team list <teammate id=" "> of team participant(s) in coached conferencing. Used with the ID attribute.

**Parents:** configure\_team, configure\_leg

## MSCML Attributes

This section describes the MSCML conferencing attributes (listed in alphabetical order).

### action

Operation to be performed on the team list.

**Attribute of:** configure\_team

#### Values

Value	Description
add	Adds a teammate.
delete	Deletes a teammate.
query (default)	Returns the teammate list.
set	Creates a team list when followed by <teammate id="n"> and also removes all the teammates from the team list, for example, when the creator (originator) of the team list on that specific conference leg wants to remove all of the teammates from the team. If the set operation removes all teammates from a participant, that participant hears the full conference mix.

### dtmfclamp

Whether the audio from the leg is removed (clamped) from the conference mix when a DTMF digit is detected.

In a conference leg with DTMF clamping set to no, detected 2833 events are re-generated (mixed) to all other conference participants. To receive a 2833 event payload, you must first set up a channel for 2833 mode.

Transmitting 2833 events through conferences is implemented with the concept of a single 2833 talker; only one leg can send 2833 at a time. The leg that first detects 2833 and is not clamped becomes the 2833 talker and has that 2833 event transmitted to completion. Once that event completes, another leg

can become the 2833 talker. This scheme prevents mixing of simultaneous 2833 events from different sources thereby confusing the recipients. However, it allows for any leg to send 2833 in the conference, but not simultaneously.

**Attribute of:** `configure_leg`

**Values:**

Value	Description
yes (default)	Audio from leg is removed when a DTMF digit is detected. The default should be modified by applications only on rare occasions.
no	Detection of DTMF does not clamp the audio from this leg.

### id

Alphanumeric characters that uniquely identify the three participants in coached conferencing. Number of characters cannot exceed 64.

**Attribute of:** `configure_team`

**Example:** `id=supervisor1`

### mixmode

Mode for mixing audio from and to this leg.

**Attribute of:** `configure_leg`

**Values:**

Value	Description
full (default)	Audio is mixed, but not as the loudest leg.
preferred	The input audio from a leg with mixmode set to preferred is assigned a higher weight than legs set to full when determining who is included in the mix. With reasonable input levels from conference participants, this setting causes the preferred leg to be part of the mix. This setting overrides the default selection of mix inputs based strictly on loudness.
parked	Complete isolation from conference input and output. Used to isolate user audio input and output so IVR operations can occur. After IVR operation is complete, returns the user's mixmode to previous setting with an additional <code>&lt;configure_leg&gt;</code> request.
private	The leg's audio input is sent only to its teammates rather than to the full conference. The participant still hears the regular conference audio.

Value	Description
mute_video	Suppresses only the video feed of the participant and allows the audio to be fed into the conference mixer. If the leg with mute_video variation is the loudest talker, then no video is sent to the other participants.
mute_audio	Suppresses only the audio from the participant and allows the video to be received from a participant.
lecture	Makes that leg the lecturer for the conference. All non-lecture participants have the audio and video muted.
lecture_video	Makes that leg the video lecturer for the conference. All non-lecture participants have the video muted once any participant becomes the lecturer_video.
lecture_audio	Makes that leg the audio lecturer for the conference. All non-lecture participants have the audio muted once any participant becomes lecture_audio.

## repeat

Provides the client application with a way of repeating tone and prompts for a given number of times with the ability to limit the duration to a given length of time, if desired. The user can set the duration for an infinite length of time, or the repeating can be terminated by an interrupt.

**Attribute of:**            prompt

**Values:**                0 - infinite (Default: 1.)

## report

Enables and disables notification for events. Report is an attribute of activetalkers, signal, and keypress.

**Attribute of:** activetalkers, signal

**Values:**

Value	Description
yes	Enables reporting of the associated event.
no (default)	Disables reporting of the associated event.

**Attribute of:** keypress

**Values:**

Value	Description
normal	Reports normal digits.
long	Reports long digits.
both	Reports both manual and long digits.
none (default)	Disables asynchronous DTMF reporting.

## reserveconfmedia

Allocates a control leg resource for playing or recording audio to or from the entire conference.

**Attribute of:** [configure\\_conference](#)

**Values:**

Value	Description
yes (default)	Must be set to yes for the application to play to or record the conference. The yes value consumes one of the available talker ports (as set by reservedtalkers) for the conference and sets the mixmode for the virtual conference leg to preferred, which ensures that conference announcements are mixed into the output as the loudest audio.
no	Does not play to or record the conference.

## reservedtalkers

How many talker legs are allocated when the conference is created.

Audio from talker legs is processed by the mixing algorithm to produce the conference output.

This attribute can be carried in the initial SIP INVITE that creates the conference or in mid-call INFO request to resize an existing conference.

---

**Note:** The `reservedtalkers` attribute is currently ignored by the IP Media Server, so the conference size is not limited by the value of `reservedtalkers`.

---

**Attribute of:** [configure\\_conference](#)

Values are numbers in the range of 2 to available media sever capacity.

## toneclamp

Removes (clamps) audio when non-voice signals are present in the media stream.

**Attribute of:** [configure\\_leg](#)

**Values:**

Value	Description
yes (default)	Audio is clamped when tones are detected.
no	Audio is not clamped.

## type

Attribute of `config_leg` and `signal`.

The type of participant for this conference leg.

**Attribute of:** [configure\\_leg](#)

**Values:**

Value	Description
talker (default)	The audio input from the participant is processed by a mixing algorithm to create the conference output.
listener	Audio input from the participant is not included in the conference mix
internal	A conference leg used for playing to or recording from a conference.

**Attribute for:** signal (required)

Identifies the type of signal to be reported.

**Values:**

Value	Description
busy	Far end busy tone (also fast busy)
dial	Dial tone
CED	Fax calling tone
CNG	Fax called station identifier
PVD_PAMD	Registering for the PVD_PAMD signaling detection registers the application for both Positive Voice Detection (PVD) and Positive Answering Machine Detection (PAMD). These do not need to be turned off. The detector runs until either PVD or PAMD is detected and then the PVD_PAMD detection is automatically disabled.  An MSCML application does not need to stop the call progress signal detection prior to the completion of the call.
PVD	The IP Media Server sends notification regarding the PVD. Detection of the event stops the detector.
PAMD	The IP Media Server sends notification regarding the PAMD. Detection of the event stops the detector.
interrupt	Call interrupted (call waiting)
ring	Far end ring tone
400	tone used in Japan



## 5 - IVR with MSCML

This chapter explains the IP Media Server's implementation for Interactive Voice Response (IVR) using the `ivr` service indicator.

The IP Media Server offers two interfaces for those choosing to develop IVR/DTMF applications: IVR with MSCML and VoiceXML. For more information about VoiceXML, see [“VoiceXML Version 2.0 and Dialog Service” \(page 182\)](#), and [“VoiceXML Version 1.0 and Dialog Service” \(page 319\)](#).

This chapter includes the following major sections:

- ◆ [IVR Service](#)
- ◆ [Playing Announcements](#)
- ◆ [Collecting DTMF Digits](#)
- ◆ [Recording Audio](#)
- ◆ [Stopping an IVR Request in Progress](#)
- ◆ [MSCML IVR Reference](#)

## IVR Service

The IVR service (ivr) supports basic Interactive Voice Response functions, such as:

- ◆ Playing announcements
- ◆ Collecting DTMF digits
- ◆ Recording audio

These functions are based on MSCML elements that are added to the message body of a SIP request.

### Example 16. IVR Service Indicator with MSCML Elements

```
To: sip:ivr@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407027 INFO
Content-Length: 237
Content-Type: application/mediaservercontrol+xml
Via: SIP/2.0/UDP 192.168.1.126:5060
<?xml version='1.0'?>
<MediaServerControl version='1.0'>
  <request>
    <playcollect barge='yes' cleardigits='yes'
      maxdigits='4'>
      <prompt>
        <audio url='http://192.168.1.126:8013/app/
          audio/askauthcode.raw' />
      </prompt>
    </playcollect>
  </request>
</MediaServerControl>
```

```
Response to INFO request from the IP Media Server:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
From: sip:appserver@192.168.1.126;tag=oi32zw
To: sip:ivr@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407027 INFO
```

The request payload (block or stream of data/information) for IVR can be located in either the initial SIP INVITE or in INFO requests.

Your application must use the INFO method for mid-call requests.




---

**The application server must support sequenced INFOs per draft-ietf-sip-rfc2543bis-03 “SIP: Session Initiation Protocol”.**

---

The IP Media Server notifies the application that the command has completed by sending a response message containing final status information and data such as collected DTMF digits. See [“IVR Response Elements and Attributes” \(page 178\)](#) for details on IVR responses.

IVR requests are not queued. If the application receives a request while another is in progress, the first operation is stopped and the new request is carried out. The IP Media Server generates a `<response/>` message for the first request and returns any data collected up to that point. If an application stops a request in progress, but does not initiate another operation, the IP Media Server issues a `<stop>` request and generates a `<response/>` message.

The IP Media Server treats a SIP re-INVITE with Hold media (`c=0.0.0.0`) as an implicit `<stop/>` request, immediately terminating the running `<play/>`, `<playcollect/>` or `<playrecord/>` request and sending a `<response/>` to the running request indicating `reason=stopped`. See IP Media Server [“IP Media Server Behavior When Hold Media is Presented” \(page 91\)](#) for additional information on Hold media.

The IVR service indicator is `ivr`.

```
INVITE sip:ivr@MS_IP
```

There are five MSCML directives for IVR functions:

- ◆ `<play/>`
- ◆ `<playcollect/>`
- ◆ `<playrecord/>`
- ◆ `<stop/>`
- ◆ `<managecontent/>`

The MSCML reference for these directives begins on page [126](#). See [“MSCML Schema” \(page 367\)](#) for the complete MSCML grammar.

See [“Sample Code and Call Flows” \(page 241\)](#) for code samples and call flow diagrams.

## Call Progress Tones

The IP Media Server supports the following call progress tones in MSCML to allow call centers to quickly classify calls as successful or not.

- ◆ `busy`
- ◆ `dial`
- ◆ `CED`
- ◆ `CNG`
- ◆ `PVD_PAMD`
- ◆ `interrupt`
- ◆ `ring`

## ◆ 400

---

Note: These call progress tones in MSCML can also be used in the IP Media Server's conferencing implementations.

---

Third party systems can subscribe for these call progress events early in the call setup to quickly detect any of the call progress signal/tones.

The type attribute value of PVD\_PAMD is used to register for both PVD and PAMD notification events.

The following is an example MSCML subscription for PVD and PAMD notification events.

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg>
      <subscribe>
        <events>
          <signal type="PVD_PAMD" report="yes"/>
        </events>
      </subscribe>
    </configure_leg>
  </request>
</MediaServerControl>
```

The following is an example MSCML signal notification.

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <notification>
    <signal type="PVD"/>
  </notification>
</MediaServerControl>
```

---

# Playing Announcements

To play an announcement without interruption and with no digit collection, use the <play/> request. For example, use this request to announce the name of a new participant to the entire conference.

To immediately halt the current request and trap any content retrieval errors, use the Stop on Error feature. For further details, see [“Handling of Content Retrieval Errors” \(page 163\)](#)

## Elements and Attributes

In the body of the request, the <prompt/> element specifies the announcement to play.

The attribute `id` (optional) is an application-defined request identifier that correlates the asynchronous response with its original request and echoes back to the application in the IP Media Server's response.

The `prompturl`, `promptencoding`, and `offset` attributes have been deprecated. Applications should use the <prompt/> element instead.

For further details concerning IVR attributes, see [“TVR Attributes” \(page 168\)](#).

## Responses

When the announcement has finished playing, the Dialogic® IP Media Server sends a <response> payload to the application in a SIP INFO message.

The response contains the following with some examples noted in parenthesis:

- ◆ Return `code` (200)
- ◆ Return `text` (ok)
- ◆ Reason (EOF)
- ◆ Actual duration of the prompt
- ◆ ID, if one was provided

For more information about responses to <play>, see [“IVR Response Elements and Attributes” \(page 178\)](#).

## Collecting DTMF Digits

To collect DTMF digits, use the `<playcollect/>` request.

This request has multiple attributes, all of which are optional. For a detailed description of the attributes, see [“MSCML IVR Reference” \(page 165\)](#).

### Prompting

The presence or absence of the `<prompt/>` element controls whether to initiate an announcement or only collect digits. The following is an example of how the `<prompt/>` element works.

```
<prompt baseurl="file:///opt/snowshore/prompts/conf/">
  <audio url="please_enter.wav"/>
  <variable type="silence" value="1"/>
  <audio url="your.wav"/>
  <variable type="silence" value="1"/>
  <audio url="pin_number.wav"/>
</prompt>
```

For additional information about the prompt element, see [“prompt” \(page 125\)](#).

### Digit Buffering

The ivr service automatically detects and buffers DTMF digits. When the IP Media Server receives a `<playcollect/>` request, it begins to examine the DTMF digits buffer.

To determine whether immediate action is required, the IP Media Server compares any previously buffered digits for a match with the following attributes:

- ◆ returnkey
- ◆ escapekey
- ◆ maxdigits

This examination provides the type-ahead behavior for menu traversal and other types of IVR interactions.

The application can override type-ahead behavior by setting the `cleardigits` attribute to `yes`, removing all previously-buffered digits. As a result, the application only considers user input that occurs after the request.



---

**Caution: If `cleardigits` is set to `no`, previously-buffered digits may result in the prompt being interrupted (barged into) immediately. As a result, prompt play does not begin and digit collection starts immediately.**

---

The default for `barge` is `yes`. If the `barge` attribute is set to `no`, `cleardigits` is implicitly understood to have a value of `yes`. This value provides that the application clears any DTMF input occurring before the current collection after the request completes.

## Star and Pound Keys

The application can set two special keys to a single DTMF digit to invoke special processing when detected.

The `escapekey` (defaults to `*`) indicates that the user intends to terminate the current operation without saving any input collected up to that point. Detection of the specified `escapekey` digit terminates the request immediately and generates a response.

The `returnkey` (defaults to `#`) indicates that the user has completed the input and wants to return all collected digits to the application. If the IP Media Server detects a `returnkey`, the IP Media Server immediately terminates collection and returns the collected digits to the application in the `<response>` message.

The IP Media Server allows you to trap any content retrieval errors and immediately halt the current request by using the `stop_on_error` attribute. For further details, see [“Handling of Content Retrieval Errors”](#) (page 163).

## Timing Attributes

Several attributes control how long the IP Media Server waits for digits in the input sequence. All timer values are expressed in milliseconds. The timers include the following:

- ◆ `firstdigittimer` controls how long the IP Media Server waits for the initial DTMF input.
- ◆ `interdigittimer` controls how long the IP Media Server waits between DTMF inputs.
- ◆ `extradigittimer` controls how long the IP Media Server waits for additional user input after the specified number of digits (`maxdigits`) have been collected.

The `extradigittimer` attribute enables the `returnkey` input to be associated with the current collection. For example, if `maxdigits` is set to 3 and the `returnkey` is set to `#`, the user can enter `x#`, `xx#`, or `xxx#`, where `x` represents a DTMF digit.

If the IP Media Server detects the `returnkey` pattern during the `extradigit` interval, the collected digits are returned to the application and as a result, the `returnkey` is removed from the digit buffer.

If this were not the case, and considering the above example, the application would receive xxx and leave the terminating # in the digit buffer to be processed by the next <playcollect/> request. This could result in unintentionally terminating the prompt that immediately follows.

---

Note: The extradigittimer has no effect unless returnkey has been defined.

---

## Responses

When the <playcollect/> has finished playing, the IP Media Server sends a <response> payload to the application in a SIP INFO message.

The <response> consists of the following, with some examples noted in parenthesis:

- ◆ Return code (200)
- ◆ Return text (OK)
- ◆ Reason (timeout)
- ◆ Actual duration of the prompt (playduration)
- ◆ Collected digits
- ◆ ID, if one was provided (application-defined)

For further details, see [“IVR Response Elements and Attributes”](#) (page 178).



---

# Recording Audio

To record an audio request, use the `<playrecord/>` request.

The `<playrecord/>` request directs the IP Media Server to capture a real-time audio stream (Real-Time Transport Protocol) and deliver the resulting content to a URL specified by the controlling application. You can only specify file scheme URLs (NFS transport).

## Playrecord Attributes

This request directive contains multiple attributes. The `recurl` attribute is required, because it identifies the URL for the recorded audio. All other attributes are optional.

The presence or absence of the `<prompt>` element controls whether or not a prompt plays before recording begins.

## Playrecord Process

When the IP Media Server requests a prompt, `<playrecord/>` has two stages.

The first is essentially a `<playcollect/>` operation. The application can set the prompt phase to be interrupted by DTMF input and can also specify an `escapekey` that will terminate the `<playcollect/>` request before the recording phase begins. The `escapekey` applies only to `<playcollect/>` and has no effect once recording starts.

When the `escapekey` is pressed, the IP Media Server generates a response message and the operation ends immediately. If any other keys are pressed and if the prompt has been set as interruptible (`barge=yes`), play stops immediately and the recording phase begins.

Any digits collected in the prompt phase, with the exception of those specified by the `recstopmask` attribute, are buffered and returned in the response.

If the request proceeds to the recording phase, the application discards any digits from the collection phase from the buffer to eliminate unintended termination of the recording.

The application compares digits detected during the recording phase to the digits specified in the `recstopmask` to determine if the digits should terminate recording.

Digits not present in the `recstopmask` are ignored and passed into the recording. If the recording is terminated because of a DTMF input, the collected digits are returned to the application in the `<response>`.

Once recording begins, the application writes the audio to the specified `recurl` URL, no matter what DTMF events are detected. The application must examine the DTMF input returned in the `<response>` message to determine whether the audio file should be saved or deleted and then re-recorded.

You have the option of trapping any content retrieval errors and immediately halting the current request by using the `stop_on_error` attribute. See [“Handling of Content Retrieval Errors” \(page 163\)](#) for more information.

## Timing Attributes

The `initsilence` and `endsilence` timing attributes control how long the IP Media Server waits for the start of speech to begin the recording and the absence of speech to end the recording.

Table 42 describes these timing attributes.

Table 42. Timing Attributes

Attribute	Definition
<code>initsilence</code>	How long to wait for initial speech input before terminating (canceling) the recording. This attribute can be a positive integer value in milliseconds, or can be set to -1, which directs the IP Media Server to wait indefinitely.  The default is 3000 ms (3 seconds).
<code>endsilence</code>	How long the IP Media Server waits after speech has ended to stop the recording. This parameter takes a positive integer value in milliseconds, or can be set to -1. With a value of -1, the recording continues indefinitely after speech has ended, but may terminate due to a DTMF keypress or because the maximum desired duration has been reached. The default maximum duration is 4000 ms (4 seconds). If the <code>endsilence</code> timer expires, the Dialogic® IP Media Server trims the end of the recorded audio by an amount equal to the <code>endsilence</code> parameter.

## Additional Attributes

Table 43 describes additional attributes associated with `<playrecord>`.

Table 43. Additional Attributes of PlayRecord

Attribute	Definition
<code>mode</code>	Whether the recording overwrites or appends.
<code>recencoding</code>	Specifies the content codec to use.
<code>duration</code>	Specifies the time in ms for the entire recording.
<code>beep</code>	Specifies whether a beep signifies the start of recording.

For details on all IVR attributes, see [“IVR Attributes” \(page 168\)](#).

## Responses

When the IP Media Server has finished recording, it generates a <response> message and sends it to the application in a SIP INFO message.

The response contains the following with some examples noted in parenthesis:

- ◆ Return code (200)
- ◆ Return text (ok)
- ◆ Reason (endsilence)
- ◆ Actual duration of the prompt (playduration)
- ◆ Collected digits
- ◆ ID, if one was provided (application-defined)

For further details, see [“IVR Response Elements and Attributes” \(page 178\)](#).

## Handling of Content Retrieval Errors

Applications can control the IP Media Server's behavior when a content retrieval error occurs and capture detailed information regarding what happened. This feature is controlled through the `stop_on_error` attribute, which pertains to the <prompt/> tag. The <prompt/> tag appears in <play/>, <playcollect/>, and <playrecord/> requests. This attribute takes the following values:

Value	Description
yes	The IP Media Server stops the request if a fetch error occurs.
no (default)	Original MSCML behavior; fetch errors do not stop the request. For compatibility reasons, this is the default.

Details of the issue are returned in the <error\_info/> element in the MSCML response. This element has three attributes: `code`, `text` and `context`. The `code` and `text` attributes are similar to their counterparts in the MSCML response; however, these attributes contain the error code and text received from the content server. The `context` attribute holds the URL that the IP Media Server was attempting to retrieve when the error occurred.

## Stopping an IVR Request in Progress

The application issues a <stop/> request to stop a request in progress and not initiate another operation. This request generates a <response/> message from the IP Media Server.

### ID Attribute

The id attribute is application-defined. This optional attribute is the only attribute related to the <stop> request.

This application-defined request ID correlates the asynchronous response with its original request and echoes back to the application through the IP Media Server's response.

### Response

The response carries the following:

- ◆ Return code (200)
- ◆ Return text (OK)
- ◆ ID, if one was provided (application-defined)

# MSCML IVR Reference

This section lists the MSCML elements and attributes that support the IVR service (ivr).

For the complete MSCML grammar, see [“MSCML Schema” \(page 367\)](#).

## IVR Elements

Table 44 summarizes MSCML elements that support IVR.

Table 44. MSCML Elements for IVR

Element	Attributes	Response Attributes	Description
managecontent	src, dest, action, httpmethod, name, mimetype, fetchtimeout	N/A	Moves recorded content from the IP Media Server to remote locations using the HTTP protocol. This is a store and forward model, which requires the local temporary recording to be completed before it can be sent to the Web server (i.e., <playrecord/>). There is no provision for streaming the recording to the Web server while it is being recorded.
play	delay, duration, id, locale, offset, prompt, promptencoding, repeat, stop_on_error	code, id, playduration, reason, text	Plays an announcement that cannot be interrupted by DTMF input and does not require digit collection. Specifies the announcement to be played by creating a <prompt> element in the body of the request.
playcollect	barge, cleardigits, delay, duration, escapekey, extradigittimer, firstdigittimer, id, interdigittimer, locale, maskdigits, maxdigits, offset, prompt, promptencoding, repeat, returnkey	code, digits, id, playduration, reason, text	Collects user input or DTMF digits when an announcement might be interrupted. The announcement is optional, and if not set, the result is digit collection only.

Table 44. MSCML Elements for IVR (continued)

Element	Attributes	Response Attributes	Description
playrecord	barge, beep, cleardigits, delay, duration, endsilence, escapekey, id, initsilence, locale, mode, offset, prompt, promptencoding, recencoding, recstopmask, recurl, repeat	code, digits, id, playduration, reason, reclength, text	Request used when recording audio to set the properties the IP Media Server uses to capture RTP and deliver it to a URL specified by the controlling application. The application can specify an optional prompt to play prior to the recording.
stop	id	code, id, text	Stops a request in progress and does not initiate another operation.

## IVR Prompt Block

The prompt block in the body of the play, playcollect, or playrecord request contains one or more references to physical audio files, provisioned sequences, or attributes that are played in the order in which they appear.

**Note:** The play queue that can be encased within the <prompt> and </prompt> keywords can only be 16 files deep. If you send a play request with more than 16 files, only the first 16 are played and a successful return code is generated.

**Attributes:** baseurl, locale, stop\_on\_error, repeat, duration, offset, delay

The following examples (17–19) illustrate several uses of the prompt block.

### Example 17. Prompt Block

```
<prompt baseurl="file:///opt/snowshore/
  prompts/conf/">
  <audio url="please_enter.wav"/>
  <variable type="silence" value="1"/>
  <audio url="your.wav"/>
  <variable type="silence" value="1"/>
  <audio url="pin_number.wav"/>
</prompt>
```

Using the repeat, duration, delay and offset attributes, the application can repeat prompts for a given number of times with the ability to limit the duration to a given length, if desired. The user can set the duration attribute for an infinite length of time or the application can terminate repetition using an interrupt <stop/>. The following examples illustrate these attributes.

**Example 18.** Prompt Block with Repeat for Single File

```

<request>
<play id=212>
  <prompt repeat="infinite" duration="infinite"
    offset="1" delay="10">
    <audio url="file:///opt/snowshore/prompts/
      generic/10.ulaw"/>
  </prompt>
</play>
</request>

```

**Example 19.** Prompt Block with Repeat for Several Files

```

<request>
<play id=21212>
  <prompt baseurl="file:///opt/snowshore/prompts/
    generic/" locale="en_US" repeat="infinite"
    duration="infinite" offset="0" delay="2000">
    <audio url="num_changed.wav"/>
    <audio url="new_number.wav"/>
    <variable type="dig" subtype="ndn"
      value="9725551212"/>
    <audio url="make_note.wav"/>
  </prompt>
</play>
</request>

```

**Prompt Elements**

The VoiceVML prompt element takes two elements to specify the file to be played: <audio> and <variable>.

**<audio>**

Each audio element in a <prompt/> block refers to a physical audio file or sequence to be played. Audio files are played in the order in which they are listed in the block.

**Attributes:** encoding, url, src

**<variable>**

A symbol or string of symbols whose value changes.

**Attributes:** subtype, type, value

## IVR Attributes

Attributes are listed in alphabetical order. All attributes are optional unless otherwise noted.

### barge

Whether a prompt gets interrupted when any DTMF digit is detected.

**Attribute for:** playcollect, playrecord

**Values:**

Value	Description
yes (default)	DTMF digit detection interrupts the prompt. <hr/> <b>Note:</b> Any digit barges in. There is no barge mask. <hr/>
no	DTMF digit does not interrupt prompt. Any digits that arrive during the prompt are discarded. The digits are not saved for processing during the collection portion of playcollect.

### baseurl

Base URL to be placed in front of the url attribute of any <audio/> tags that do not match a supported scheme (`file:///`, `http://`). This is a notational convenience to reduce the size of the MSCML payload.

**Attribute for:** prompt

### beep

Whether a beep is played to signify the start of a recording.

**Attribute for:** playrecord

**Values:**

Value	Description
yes (default)	The IP Media Server plays a tone to signify the start of a recording. The beep is an 880 Hz tone 250ms in duration.
no	No tone is played at the start of a recording.

### cleardigits

Whether the application removes previously collected digits from the buffer before the command starts. For type-ahead, set cleardigits to no.



**Attribute for:** playcollect, playrecord

**Values:**

Value	Description
yes	Digits are flushed.
no (default)	Digits are not flushed.

## delay

Time between plays for the entire sequence (in milliseconds).

**Attribute for:** prompt

Values range from 0 – infinite. (Default: 0).

## duration

Number of milliseconds that the entire recording can span. Recordings continue until an ending condition (such as a digit or ending silence) is detected or until the recording media is full.

**Attribute for:** playrecord

**Values:** 0 - infinite (Default: infinite.) A value of -1 means no limit is imposed.

## encoding

Content encoding of physical audio files that are not in self-describing .wav or .au formats.

**Attribute for:** audio

**Values:**

Value	Description
µlaw	G.711 µlaw encoding.
alaw	G.711 alaw encoding.

There is no default.

## endsilence

Number of milliseconds of trailing silence that can elapse before terminating the recording. The IP Media Server reduces the recording by an amount equal to the endsilence value if endsilence is set.

**Attribute for:** playrecord

**Values:** a number in milliseconds. (Default: 4000 ms.) To disable the setting, use a value of -1, which indicates that no endsilence timer should be applied.

### **endwarning**

Number of milliseconds at the end of the recording to play warnurl.

**Attribute for:** playrecord

**Values:** a number in milliseconds.

### **escapekey**

A digit that is used to terminate the current operation.

When this digit is pressed, the command terminates and the digit buffer is cleared. No digits are returned to the application, and the reason escapekey is returned with the <response/> event.

If detected in the play phase, the prompt stops immediately, and the command completes without entering the record phase.

**Attribute for:** playcollect, playrecord

**Values:**

Default is \* (star key). If no escapekey is desired or if you want to collect \* as a digit, you can set escapekey to be empty using two consecutive double quotes, for example, escapekey"".

### **extradigittimer**

Amount of time allowed for the user to enter an additional digit after the desired number of digits has been collected. This attribute is typically used when collecting variable-length inputs with a returnkey parameter defined. If this timer elapses, the command completes with the reason=timer.

**Attribute for:** playcollect

**Values:** A number in milliseconds.(Default: 1000 ms.) A value of -1 directs the IP Media Server to wait indefinitely for additional input.

---

**Note:** If the extra digit detected is the returnkey, then all collected digits are returned.

---

### **firstdigittimer**

Amount of time allowed for the user to enter the first DTMF digit. If the timer elapses, the command completes with the reason nodigits.

**Attribute for:** playcollect

**Values:** A number in milliseconds. (Default: 5000 ms.) A value of -1 directs the IP Media Server to wait indefinitely for the first input. Using this setting for conferences ensures that user input is not ignored.

## id

An application-defined request identifier that correlates the asynchronous response with its original request. The provided ID is returned back to the application in the IP Media Server's response.

Optional.

**Attribute for:** play, playcollect, playrecord, stop

## interdigittimer

Amount of time allowed for the user to enter additional digits before the desired number of digits have been collected. If this timer elapses, the command completes with the reason timer.

**Attribute for:** playcollect

**Values:** A number in milliseconds. (Default: 2000 ms.) A value of -1 directs the IP Media Server to wait indefinitely for additional input.

## initsilence

Number of milliseconds of initial silence that can elapse before the recording is terminated. If speech is detected in this interval, recording proceeds.

**Attribute for:** playrecord

**Values:** A number in milliseconds. (Default: 3000ms.) A value of -1 indicates that no initial silence period is desired or alternately that the IP Media Server should wait forever for the start of speech. The value of -1 effectively disables the setting.

## locale

Language and country variant to be used when resolving variables contained in the <prompt> block. The language is defined as a two-letter code per ISO 639. The country variant is also defined as a two-letter code per ISO 3166. These elements are concatenated with a single underscore (%x5F) character.

**Attribute for:** prompt

**Example:** locale=en\_US

## maskdigits

Manages clear text logging.

**Attribute for:** playcollect, event report, report keypress

**Values:**

<b>Value</b>	<b>Description</b>
Yes	Sensitive data is not be logged in clear text. DTMF data that is normally written to the IP Media Server log files is masked or discarded.  Masked DTMF data appears as a '-' character in the log files. Discarded DTMF is indicated by the text string <?Sensitive Data?> in the log files.
No (default)	DTMF data is recorded and written to the IP Media Server log files.

## maxdigits

Maximum number of digits to be collected and returned by the IP Media Server.

**Attribute for:** playcollect

**Values:** 1–62. (Default: 1.)

## mode

Whether the recording overwrites or appends to the supplied URL.

---

**Note:** This attribute only affects file:/// scheme URLs.

---

**Attribute for:** playrecord

**Values:**

Value	Description
overwrite (default)	Recording overwrites the contents of the URL.
append	Recording is appended to the contents of the URL.

## offset

Position in a prompt or sequence where play begins. The value of this attribute is expressed in milliseconds with 0 indicating the start of the prompt or sequence.

Offset can be either outside or inside the prompt block. The offset outside the prompt block supports existing code and refers to the offset in a single set. If the offset is included inside the prompt block, it works the same way, but only on the first set of repeats.

The IP Media Server supports both positive (skip forward) and negative (skip back) values. Offsets apply to sequences as well as files. If the specified offset cannot be reached in the current segment, the next one is processed until the desired offset is reached.

**Attribute for:** prompt, play, playcollect, playrecord

**Values:** 0 - infinite (Default: 0.)

## promptencoding

Encoding of physical audio files that are not in self-describing .wav or .au formats.

---

**Note:** This attribute only affects file:/// scheme URLs.

---

When using a physical audio file, the encoding a-law or  $\mu$ -law must be specified in the file header, in the extension, or by using this attribute.

**Attribute for:** play, playcollect, playrecord

**Values:**

Value	Description
ulaw	G.711 $\mu$ -law encoding.
alaw	G.711 a-law encoding.
msgsm	MSGSM

If not specified, the promptencoding is determined from either the file extension, or if the first four bytes of the file indicate the presence of a RIFF or .snd header.

## recstopmask

DTMF digits that terminate the recording. Input digits not contained in the mask are ignored and passed into the recording. The terminating DTMF digit is returned to the application in the <response/> event.

**Attribute for:** playrecord

**Values:** 0 1 2 3 4 5 6 7 8 9 \* #

## recurl

Target for the recording. Only NFS (file:///) is supported for recording. Filenames ending in .au or .wav produce files written in those formats. Required.

**Attribute for:** playrecord

## recencoding

Encoding used for the recording.

**Attribute for:** playrecord

**Values:**

Value	Description
ulaw (default)	G.711 $\mu$ -law encoding.
alaw	G.711 a-law encoding.
ms_gsm	MSGSM

**repeat**

Provides the client application with a way of repeating tones and prompts for a given number of times with the ability to limit the duration to a given length of time, if desired. The user can set the duration for an infinite length of time or the repeating can be terminated by an interrupt <stop/>.

**Attribute for:** prompt

**Values:** 0 - infinite (Default: 1.)

---

**Note:** The value infinite means the prompt should be repeated indefinitely.

---

**report**

Enables or disables notification for events. Report is an attribute for both signal and keypress.

**Attribute for:** signal

**Values:**

Value	Description
yes	Enables reporting of the associated event.
no (default)	Disables reporting of the associated event.

**Attribute for:** keypress

**Values:**

Value	Description
normal	Reports normal digits.
long	Reports long digits.
both	Reports both manual and long digits.

Value	Description
none (default)	Disables asynchronous DTMF reporting.

## returnkey

Digit that signifies the end of the collection process.

When this digit is pressed, the command terminates. All collected digits are sent to the application, and the reason `returnkey` is returned in the `<response>` event with the collected digits.

**Attribute for:** `playcollect`

**Values:** 0 1 2 3 4 5 6 7 8 9 \* # A, B, C, D

The default is # (the pound key).

If the `returnkey` is not required or if you want to collect # as a digit, set the `returnkey` to be empty by using two consecutive double quotes for example, `returnkey=""`.

## stop\_on\_error

Controls IP Media Server handling and reporting of errors encountered when retrieving remote content.

**Attribute for:** `prompt`

**Values:**

Value	Description
yes	The IP Media Server stops the request if a fetch error occurs.
no (default)	Original MSCML behavior; fetch errors do not stop the request. The default is no for compatibility reasons.

## subtype

Minor or subtype values describing how to render a variable as audio.

**Attribute for:** `variable`

**Values:**

Value	Spoken as...
mdy 20021015	"October Fifteenth Two Thousand Two"
ymd 20021015	"Two Thousand Two October Fifteen"



Value	Spoken as...
ndn	North American dialing phone number phrasing (NPA-NXX-XXXX), with appropriate pauses
dmy	20021015 is spoken as “Fifteen October Two Thousand Two”.
t12 1700	“Five p.m.”
t24 1700	“Seventeen hundred hours”
gen	generic digits (for example: “one”, “five”, “zero”)
crd	“one”
ord	“first”

For detailed descriptions of subtype values, see [“Variable Types and Subtypes” \(page 102\)](#).

## type

Type is an attribute for signal and variable.

**Attribute for:** signal (required)

Identifies the type of signal to be reported.

**Values:**

Value	Description
busy	Far end busy tone (also fast busy)
dial	Dial tone
CED	Fax calling tone
CNG	Fax called station identifier
PVD_PAMD	Registering for the PVD_PAMD signaling detection registers the application for both Positive Voice Detection (PVD) and Positive Answering Machine Detection (PAMD). These do not need to be turned off. The detector runs until either PVD or PAMD is detected and then the PVD_PAMD detection is automatically disabled.  An MSCML application does not need to stop the call progress signal detection prior to the completion of the call.
PVD	The IP Media Server sends notification regarding the PVD. Detection of the event stops the detector.

Value	Description
PAMD	The IP Media Server sends notification regarding the PAMD. Detection of the event stops the detector.
interrupt	Call interrupted (call waiting)
ring	Far end ring tone
400	tone used in Japan

**Attribute for:** variable (required)

Identifies the type of variable to be rendered as audio.

**Values:** date, digit, duration, money, month, number, silence, string, time, weekday.

For detailed descriptions of subtype values, see [“Variable Types and Subtypes” \(page 102\)](#).

## url

URL of the audio to be played. Can resolve to a physical file or a provisioned sequence.

**Attribute for:** audio (required)

## value

String value for conversion to audio according to its type and subtype for example, 12/25/02.

**Attribute for:** variable (required)

## warnurl

Specifies the target URL for the warning message.

**Attribute for:** playrecord

## IVR Response Elements and Attributes

The IP Media Server acknowledges receipt of an application request by sending a response of either 200 OK or 415 BAD MEDIA TYPE. (The IP Media Server sends the 415 BAD MEDIA TYPE response when the SIP request contains a content type other than application/sdp or application/mediaservercontrol+xml).

The <response> message is transported in a SIP INFO request.

If there is an error in the request or the request cannot be completed, the <response> message is sent shortly after receiving the request. If the request is able to proceed, the <response> contains final status information.

## Response Elements

The following elements are used in response messages.

### error\_info

Contains details of remote content retrieval errors, if enabled by the stop\_on\_error attribute of <prompt/>.

**Response for:** play, playcollect, playrecord

**Values:** Contains code, text, and context attributes that contain specific error information.

### id

If an application-defined ID was specified in the request, that ID is returned to the application in the response.

**Response for:** play, playcollect, playrecord, stop

## Response Attributes

### code

Status of command.

**Response for:** play, playcollect, playrecord, stop

**Values:**

Value	Description
200	Command completed.
400	For playrecord: command not accepted due to error. The text attribute describes the cause of the error.
501	For playrecord: error because the URL type specified was not supported.

### digits

Collected digits, if any.

**Response for:** playcollect, playrecord

### playduration

Elapsed play time (in milliseconds) of a prompt or sequence before the request completed due to user intervention or end of sequence.

**Response for:** play, playcollect, playrecord, stop

### reason

Cause for ending the play, playcollect, or playrecord command.

**Response for:** play

**Values:**

Value	Description
EOF	The play was completed when the end of file was reached.
stopped	A stop command, another command, or a SIP re-INVITE with hold media stopped the play.

**Response for:** playcollect

**Values:**

Value	Description
match	A digit was detected.
timeout	No digit was received in time.
returnkey	The return key terminated the operation.
escapekey	The escape key terminated the operation.
interrupted	Another request was received before the current one could finish.
stopped	A stop command, another command, or a SIP re-INVITE with hold media stopped the play.

**Response for:** playrecord

**Values:**

Value	Description
digit	A digit was detected.
endsilence	Trailing silence was detected.
init_silence	No voice (initial silence) was detected.
max_duration	Time for recording was complete.
stopped	A stop command, another command, or a SIP re-INVITE with hold media stopped the play.
escapekey	The escape key was entered either in the play mode or in the record mode. The application terminated the recording.
error	The operation failed.

**reclength**

Length of the recording in bytes.

**Response for:** playrecord

**text**

Whether the command succeeded.

**Response for:** play, playcollect, playrecord, stop

Error opening file for <playrecord> indicates requested record destination URL file could not be opened.

URL type is not implemented for <playrecord>. Indicates that the requested record destination URL file type is not supported.

## 6 - VoiceXML Version 2.0 and Dialog Service

This chapter explains the basics of VoiceXML (VXML) Version 2.0, lists the supported VoiceXML elements and attributes, and describes ECMAScript language functionality.

The dialog service is one of two interfaces that the Dialogic® IP Media Server offers for developing IVR/DTMF/Voice applications. The other service is `ivr` through which SIP requests are enhanced by MSCML message bodies for play, playcollect, and playrecord. For details on IVR, see [Chapter 5, “TVR with MSCML”](#).

This chapter includes the following sections:

- ◆ [About VoiceXML](#)
  - ◆ [VoiceXML Interpreter](#)
  - ◆ [Dialog Service Indicator and Request URI](#)
  - ◆ [VoiceXML Launcher](#)
  - ◆ [Default Script](#)
  - ◆ [VoiceXML Concepts](#)
  - ◆ [VoiceXML Application and Its Documents](#)
  - ◆ [File Storage and Retrieval](#)
  - ◆ [Media Content Recovery Extension](#)
- ◆ [VoiceXML Elements Reference](#)
- ◆ [VoiceXML Properties](#)
- ◆ [ECMAScript Functionality](#)
- ◆ [VoiceXML Extended Session Variables](#)
- ◆ [Text Overlay Functionality](#)

# About VoiceXML

VoiceXML is a W3C standard scripting language for playing text to speech and audio prompts, and for collecting DTMF and voice input.

The interpreter executes VoiceXML dialogs on an RTP stream. Each dialog represents an announcement, menu, or other IVR script. The dialogs finish when they have posted information to a Web server or returned a namelist back to the command that invoked the browser. If additional dialogs are needed, then the application runs another script.

## VoiceXML Interpreter

The IP Media Server includes interpreters for VoiceXML 1.0 and VoiceXML 2.0.

If the IP Media Server receives a SIP INVITE request directed to the dialog service, a VoiceXML session begins. The VoiceXML session initially gets (fetches) and executes the VoiceXML script that is specified in the SIP Request-URI parameter, `voicexml`.

Once specified, a VoiceXML script URI remains in effect until it completes execution or the session is stopped.

## Dialog Service Indicator and Request URI

Through the dialog service (`dialog`), the IP Media Server executes VoiceXML documents to offer IVR scripting with DTMF and voice input and recorded audio output.

The application references the initial VoiceXML script using the Request-URI parameter `voicexml`. The following SIP Request-URI directs the IP Media Server to retrieve and execute `script1.vxml` from the server `app1.carrier.com`.

```
INVITE sip:dialog@MS_IP;voicexml=http://  
  appl.carrier.com/path/  
  script1.vxml;
```

### With Resource Parameter:

```
INVITE sip:dialog@MS_IP;voicexml=http://  
  appl.carrier.com/path/  
  script1.vxml;resource=asr+tts
```

The VoiceXML 2.0 browser also supports an optional *resource* parameter. It indicates to the browser which MRCP resources are required by the VoiceXML script. It supports the values, `asr`, `tts`, and `asr+tts` to indicate if an ASR, or a TTS, or both an ASR and TTS resource are needed by the script respectively.

Providing this parameter ensures the VoiceXML browser attempts to reserve the specified resources as the call is being established as opposed to the browser requesting the resource as needed in the middle of the dialog.

It is common for the HTTP URI referencing the VoiceXML script to be a query that contains its own parameters so the script can be dynamically generated. For example, the following HTTP URI provides a subscriber ID so the VoiceXML script can be appropriately personalized.

```
INVITE sip:dialog@MS_IP;voicexml=http://
  appl.carrier.com/cgi/bin/
  genvxml.pl?subscriberid=34590087
```

If a query HTTP URI is used, be careful to replace characters that are reserved in SIP with their hexadecimal equivalents preceded by a percent (%) character. This substitution is called escaping. In particular, the question mark (?) and equals sign (=) must be escaped to conform with SIP standards. The following example shows the escaped form of the previous example.

```
INVITE sip:dialog@MS_IP;voicexml=http://
  appl.carrier.com/cgi/bin/
  genvxml.pl%3Fsubscriberid%3D34590087
```

The SIP URI must be escaped, as described in [“Syntax and Escaping” \(page 88\)](#).

## VoiceXML Launcher

If a SIP INVITE to the IP Media Server is directed to the dialog service but does not specify a file (voicexml=), then the IP Media Server launches the dialog service and runs a default script configured by the system administrator using the Web UI Media Server > VoiceXML menu.

## Default Script

The IP Media Server supports the following default script which is loaded prior to the VoiceXML script supplied in the SIP URI:

```
/opt/snowshore/htdocs/defaults.xml
```

This script provides several default properties, default <catch> blocks, and prompts that application developers can modify for their own deployment.

Please use caution when modifying the script because it affects how the VoiceXML browser behaves. Refer to the *W3C Voice Extensible Markup Language* specification for detailed information.



## VoiceXML Concepts

### Syntax

VoiceXML (VXML) is an eXtensible Markup Language for the creation of IVR and Automated Speech Recognition (ASR) applications. Based on XML tag/attribute format, its syntax involves enclosing instructions (items) within a tag structure. For example:

```
<element_nameattribute_name="attribute_value">
...contained items...
</element_name>
```

Generic XML concepts remain unchanged in VoiceXML. For example, any character data must be escaped as per the XML specification.

### Scope

As for all XML, VoiceXML observes a hierarchical structure and applies the construct of scope to define the range within the source where a variable, event handler, or other element is applicable.

### Resource Fetching

VoiceXML defines several attributes to qualify properties relevant to the caching and fetching of documents and other resources.

---

Note: The Dialogic® IP Media Server does not currently support these attributes.

---

## VoiceXML Application and Its Documents

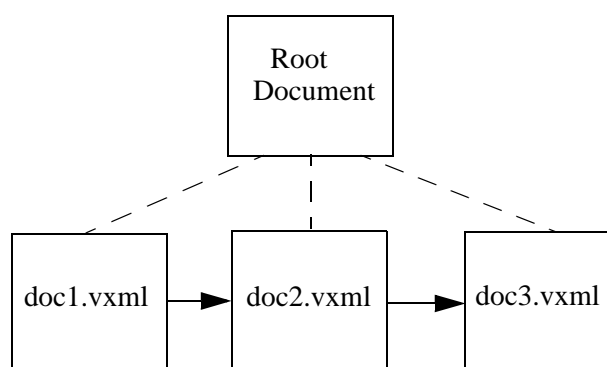
A VoiceXML application is a set of VoiceXML scripts that share the same root document. The application consists of one or more text files called documents. Document files are identified by a .vxml extension and are retrieved using HTTP, NFS (Network File System), or RTSP (Real Time Streaming Protocol).

If a VoiceXML application includes multiple documents, one of these can be the application root document. The application shares the root document among all other documents. Sharing root documents occurs when the script author explicitly sets the root document to be the same in multiple VoiceXML scripts. The IP Media Server's VoiceXML browser supports the VoiceXML standard's use of root documents.

Whenever a user interacts with a document in an application, the corresponding root document is also loaded. The root document is unloaded when a user transitions to a document not in the application.

While the root document is loaded, the application root document's variables are available to other documents as application variables. Grammars defined in the root document also remain active for the duration of the application.

Figure 7 shows how a root document is shared in a VoiceXML application.



**Figure 7. Sharing of Root Document in VoiceXML Application**

## Dialogs

A VoiceXML document contains dialogs or conversations between a prompting VoiceXML interpreter and a responding caller.

Conversational building blocks are as follows:

- ◆ Prompts (in the IP Media Server context for playing audio). If a script requires only playing recorded prompts and uses DTMF recognition, you do not need an ASR and/or a TTS resource.
- ◆ The grammar that specifies the range of acceptable user responses to a prompt
- ◆ Transitions (to the URI of the next prompt/response exchange)

Conversational constructs are as follows:

- ◆ Forms that collect an input or event, process it, and select the next form to visit.
- ◆ Menus for organization and navigation (multiple-choice options, with a transition for each).
- ◆ Links that execute a transaction or throw an event.

## Scripting

The IP Media Server supports ECMA tags that involve the scripting capabilities of the VoiceXML interpreter. See [“ECMAScript Functionality” \(page 230\)](#).

However, executing complex ECMAScript on the IP Media Server can affect its ability to perform real-time media processing. To conserve processing capacity and memory, limit the use of ECMAScripting on the IP Media Server.

## Scripting Guidelines

The following guidelines provide good practices for writing VoiceXML scripts.

- ◆ Defining grammars is the first step in the MRCP recognition process. Once you define a grammar, VXML tags such as *filled* are used to trigger the MRCP recognition processing.
- ◆ You can define grammars in the VXML script or they can be pre-compiled, stored on the MRCP server, and referenced with a URI. Pre-compiled grammar improves the performance of the application.
- ◆ Tuning the grammar is a key step in getting the best performance from a VXML script. Vendors offer commercial tools to help analyze and refine grammars.
- ◆ You can define grammar as part of the recognition command but there is a performance impact.
- ◆ Finding a grammar command in a VXML script triggers the creation of a MRCP ASR session when the VXML script is loaded and parsed.
- ◆ VoiceXML 2.0 requires that a grammar be either a voice or a DTMF grammar. A single grammar can not be both voice and DTMF. A script can have multiple grammars. VXML 2.1 will allow a single grammar to be both voice and DTMF.

## DTMF Grammars Guidelines

The IP Media Server locally supports DTMF grammars. The following are some guidelines and limitations associated with this support:

### <grammar>

- ◆ DTMF SRGS grammars must have the `type` attribute set to `application/srgs+xml` and a `mode` attribute set to `dtmf`.
- ◆ The IP Media Server supports the `semantics/1.0` value for the `tag-format` attribute of DTMF SRGS grammars.

### <tag>

- ◆ SRGS DTMF grammars can use `<tag>` elements for semantic interpretation. However only one `<tag>` element will be used for semantic interpretation for a matched grammar.
- ◆ The format of the `<tag>` content should be as follows:

```
<tag>maincourse='pizza';</tag>
```

Where `maincourse` is the variable to be filled.

### <ruleref>

- ◆ The IP Media Server does not support SRGS DTMF grammars that have external references. Specifically this means that the 'uri' attribute of the <ruleref> element is only supported if it starts with the character "#". The '#' character indicates that the 'uri' is a reference to a <rule> in the current SRGS grammar that is being parsed.
- ◆ The 'special' attribute of the <ruleref> element is not supported. Equivalent behavior can be supported with the use of the <item> element's repeat attribute.

### <rule>

- ◆ DTMF digit content as direct descendant of the <rule> element is not supported. The equivalent grammar behavior can be accomplished by having an <item> element that contains the DTMF digit content as the direct descendant to the <rule> element.

## MRCP

VoiceXML 2.0 supports Media Resource Control Protocol (MRCP) version 2.0. MRCP controls media service resources such as speech synthesizers, recognizers, and verifiers over a network. This protocol is designed to work with signaling protocol SIP which is used to establish control connections to MRCP servers. Requests are then made to the MRCP server to stream media or to perform recognition on specific media streams set to it. Media streamed both to and from the MRCP server is sent as RTP (Real Time Protocol) packets. MRCP integrates speech recognition and text to speech engines from Nuance eliminating the need for proprietary APIs. Speech recognition software applications that listen to words spoken over a telephone and recognize these words can pass the recognized words as text to a Text-To-Speech (TTS) software application that synthesizes speech from application text for playback over a telephone. Recognition and text to speech capabilities can be implemented using VXML 2.0 scripts using the Dialogic® VoiceXML 2.0 integrated with MRCP 2.0.

## Session Variables

VoiceXML provides a facility, called session variables, to pass signaling information to VoiceXML scripts. The IP Media Server supports the standard VoiceXML session variables as well as extensions which capture additional information from the SIP INVITE. See [“VoiceXML Extended Session Variables” \(page 233\)](#) for further details.

## File Storage and Retrieval

You can access VoiceXML scripts through the network using NFS and HTTP protocols. Audio and video content can be retrieved and stored using HTTP and NFS as well.

---

**Note:** The VoiceXML files can be located anywhere as long as the VoiceXML interpreter can access them.

---

## Media Content Recovery Extension

The Media Content Recovery mechanism has been extended such that there can be increased reliability of recorded content delivery from VoiceXML applications.

"Next-generation" application architectures such as SIP and VoiceXML are highly distributed in nature and rely on multiple components and network communication protocols to function. In the case of VoiceXML applications, the link between the VoiceXML browser (the IP Media Server) and the application server, which provides the scripts and content, is critical.

Now that SIP and VoiceXML applications have been successfully deployed, carriers expect reliability and behavior similar to the legacy applications that are being replaced. In most situations, it is acceptable for a subscriber to call back and establish a new VoiceXML session if a failure occurs. The notable exception is when the caller has successfully recorded a message and expects for it to be sent. If the failure occurs before the message can be transmitted, there is no way to inform the user of the issue and there is nothing the user can do about it. The Media Content Recovery extension provides a solution to the specific issue of reliable content delivery.

Recording is implemented in VoiceXML through the `<record>` element. When a `<record>` element is processed, the IP Media Server creates a temporary file with a locally unique identifier name.

The IP Media Server VoiceXML 2.0 browser maintains a list of the temporary files created during each call and deletes them when the VoiceXML script terminates. The browser also deletes any temporary files upon startup. When the Media Content Recovery mechanism is used, temporary recordings that are left over after a failure will be processed. At startup, a recovery daemon detects temporary files that have been tagged for recovery.

The Media Content Recovery extension utilizes the `<data>` element to enable the application to associate recovery data with specific recorded content. The browser supports recovery data in the form of a completely specified HTTP URI. It is the VoiceXML application's responsibility to make sure the URI is correct and contains the information needed to deliver the content to the ultimate recipient.

If the Media Content Recovery extensions are not used, the IP Media Server processes the recording in accordance with standard VoiceXML. If the VoiceXML Media Content Recovery extensions are present in the VoiceXML

script, then the recovery feature is used. A VoiceXML script that provides the user with a confirmation probably should not include the VoiceXML extensions due to the possibility that the recording could be recovered and sent without the user providing confirmation.

Refer to the [<data>](#) element for detailed information.

# VoiceXML Elements Reference

This section describes the VoiceXML elements (tags) supported by the Dialogic® IP Media Server.

## <assign>

Assigns a value to a variable.

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attributes:**

Attribute	Description
name	Name of the variable being assigned to a value. Required.
expr	ECMAScript expression that, when evaluated, is assigned as the new value of the variable. It must be a valid ECMA value. Required.

## <audio>

Plays pre-recorded audio, video or multimedia, and accesses the media file from the URI, which is specified by the src attribute.

The audio encoding is determined from the file header when .wav or .au formats are used. Otherwise, the audio encoding is implied by the file extension according to the following table.

Extension	Encoding
ulaw	G.711μ-law
alaw	G.711 A law
msgsm	Microsoft® GSM
3gp or 3gpp	3GPP
3g2 or 3gpp2	3GPP2

If the encoding cannot be determined by the extension, it is assumed to be G.711 μ-law.

The value of this tag can be empty or can contain one or more of the children listed below.

**Parents:** [<audio>](#), [<emphasis>](#), [<enumerate>](#), [<paragraph>](#), [<p>](#), [<prompt>](#), [<prosody>](#), [<sentence>](#), [<s>](#), [<speak>](#), [<voice>](#)

**Children:** [<audio>](#), [<catch>](#), [<emphasis>](#), [<mark>](#), [<paragraph>](#), [<p>](#), [<phonename>](#), [<prosody>](#), [<say-as>](#), [<sentence>](#), [<s>](#), [<sub>](#), [<voice>](#)

**Attributes:**

Attribute	Description
src	String literal containing the URI for the audio. Required.

---

Note: The pre-recorded audio identified by the “src” attribute is played using the IP Media Server resources unless it is a child of any of the listed parent to the TTS server for processing.

---

### **<block>**

Defines a container for non-interactive executable content such as welcome prompts.

The value can be empty or can contain one or more of the children listed below.

**Parent:** [<form>](#)

**Children:** [<assign>](#), [<audio>](#), [<clear>](#), [<disconnect>](#), [<exit>](#), [<goto>](#), [<if>](#), [<log>](#), [<prompt>](#), [<reprompt>](#), [<return>](#), [<script>](#), [<submit>](#), [<throw>](#), [<value>](#), [<var>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
name	Name of the form-item variable used to track whether this block can be executed. Defaults to an inaccessible internal variable.

### **<catch>**

Defines an event handler within the current scope.

The value can be empty or can contain one or more of the children listed below.

**Parents:** [<field>](#), [<form>](#), [<initial>](#), [<menu>](#), [<record>](#), [<subdialog>](#), [<transfer>](#) [<vxml>](#)



**Children:** [<audio>](#), [<clear>](#), [<data>](#), [<disconnect>](#), [<enumerate>](#), [<exit>](#), [<foreach>](#), [<goto>](#), [<if>](#), [<log>](#), [<prompt>](#), [<reprompt>](#), [<return>](#), [<script>](#), [<submit>](#), [<throw>](#), [<value>](#), [<var>](#),

**Attributes:**

Attribute	Description
event	Event or events to catch. If the attribute is not provided, all events should be caught.
count	<p>The occurrence of the event. (Default: 1.) The count allows you to handle different occurrences of the same event differently. Each form item and &lt;menu&gt; maintains a counter for each event that occurs while it is being visited; these counters are reset each time the &lt;menu&gt; or form item's &lt;form&gt; is re-entered.</p> <p>For prompts, the minimum count for the prompt to be played. Set to zero when dialog is initialized. Increases by one each time user is prompted. The &lt;clear&gt; tag resets it to zero.</p> <p>For event handlers, the minimum count for the handler to be eligible to handle an event. Set to zero at initialization and increased by one each time event is triggered.</p>
cond	Boolean value of TRUE or FALSE. (Default: TRUE). When FALSE, the tag is ignored or skipped; the form is not visited.

## <choice>

Defines a menu item by specifying one choice in the menu. It can specify a grammar fragment or the URI to go to when the choice is selected. When the choice is made, the tag either transitions to a new dialog or throws an event.

**Parent:** [<menu>](#)

**Children:** [<audio>](#), [<grammar>](#), [<say-as>](#), [<value>](#)

**Attributes:**

Attribute	Description
dtmf	DTMF sequence for the choice.
accept	<p>Overrides the setting for the accept attribute in &lt;menu&gt;. Optional. (Defaults to exact.)</p> <ul style="list-style-type: none"> <li>♦ exact - The text contained in the &lt;choice&gt; element defines the exact phrase to be recognized.</li> <li>♦ approximate - The text contained in the &lt;choice&gt; element defines an approximate recognition phrase. A subset of the words in the phrase can be matched. For example, "hello world" can be matched with "hello world", "hello", or "world".</li> </ul>

Attribute	Description
event	An event to be thrown instead of going to ext.
eventexpr	An ECMAScript expression to be evaluated and used as the name of the event to be thrown when this choice is selected. Exactly one of next, expr, event, or eventexpr must be specified; next and expr have precedence over this attribute.
fetchaudio	The URI of audio to play while waiting for the document to be fetched. Optional.
fetchhint	Defines when the next document should be fetched. Optional. <ul style="list-style-type: none"> <li>safe - only load the next document when needed.</li> </ul>
fetchtimeout	The length of time to wait for the next document to be fetched before throwing an error.badfetch event. Optional.
maxage	Indicates that this document is willing to use a cached copy of the next document only while the age of the cached copy is less than or equal to the number of seconds specified by this attribute. Optional.
maxstale	Indicates that this document is willing to use a cached copy of the next document that has exceeded its expiration time by as much as the number of seconds specified by this attribute. Optional.
message	A message string providing additional context about the event being thrown. The message will be available as a variable within the scope of the <catch> element. Optional. (Can only be used if event or eventexpr is specified; only one of message or messageexpr can be specified.)
messageexpr	An ECMAScript expression to be evaluated and used as a message string, as documented under message, above. Optional. (Can only be used if event or eventexpr is specified; only one of message or messageexpr can be specified.)
expr	Expression to evaluate instead of going to next.
next	URI of next dialog or document.

To indicate the action to take, set one of the following: next, expr, or event.

### <clear>

Resets one or more form items to its initial state, including setting the form item variable to ECMAScript undefined and re-initializing the prompt counter and the event counters for the form item.

The value of this tag must be empty.

**Parents:** [<block>](#), [<catch>](#), [<error>](#), [<filled>](#), [<help>](#), [<if>](#), [<noinput>](#), [<nomatch>](#)

**Attributes:**

Attribute	Description
namelist	List of items to clear. (When not specified, the application clears all form items in the current form.) Required.

## <content>

Specifies the content of a text overlay used in a video stream. For an example of text overlay, see [“Text Overlay” \(page 304\)](#). For more on text overlay, see [“Text Overlay Functionality” \(page 236\)](#).

**Parents:** [<overlay>](#)

**Children:** [<font>](#), [<scroll>](#)

**Attributes:**

Attribute	Description
src	Source URL of the text file. You can specify this attribute or omit this attribute and specify inline content (that is, text strings placed in the element). One or the other must be specified. For example, to use GB18030 encoding, you must specify a local file: <pre>&lt;content src="file:///opt/snowshore/htdocs/vxml/ demos/media/pizza.gb" encoding="GB18030"&gt;</pre>
type	Used with src. The MIME type of the src. Currently the only supported value is: plain/text. Default value is plain/text.
encoding	The encoding type of the text. Values are: UTF8, ASCII, or GB18030. Default value is ASCII.
halign	Horizontal alignment of the text in the overlay area. Values are: left, right, and center. Default value is left.
wordwrap	Specifies whether to wrap text to the next line in the overlay area. Values are: on or off. Default value is off.

**<data>**

Enables [Media Content Recovery Extension](#). The Media Content Recovery feature is triggered by a <data> element with either of the following set:

- ◆ The src property set to builtin:persist.  
or
- ◆ The srcexpr property evaluates to the ECMA string builtin:persist.

An error.semantic event is generated if the <data> element is encountered with src or srcexpr set to anything but builtin:persist.

An error.semantic is generated if the <data> tag has any of the following attributes set with the builtin:persist data submission:

- ◆ method, enctype, fetchaudio, fetchint, fetchtimeout, maxage, maxstale

A valid <data> tag causes the recovery file to be created. If the recovery file cannot be written, an error.semantic event is generated.

The <data> tag is controlled entirely by its namelist attribute.

The namelist attribute must have one and only one variable name that is the same as the <record> element to be protected by the Media Content Recovery feature. The <record> input item must be in the same <form> element. If it is not, an error.semantic event is generated.

The namelist attribute must have one and only one recovery\_uri variable name that evaluates to a defined ECMA value that is not an array or an object. Without the recovery\_uri variable name the recovery file will not be written. The recovery daemon attempts content recovery by executing an HTTP POST method to the HTTP URI specified in the recovery information. When a file is being recovered, the recovery daemon uses the recording\_name variable as the name of the content in the HTTP POST.

The recovery file is deleted if a new <data> element is encountered that does not have the recovery\_uri name in the namelist. This allows an application to immediately disable the recovery feature if the user confirms that he does not want the message sent.

---

**Note:** Other uses of <data> as part of VoiceXML 2.1 are not supported.

---

**Parents:**     <block>, <if>, <filled>, <form>, <noinput>

The <data> tag must be a descendant of a <form> tag that contains the associated <record> tag.

**Children:**    None.

**Attributes:**

Attribute	Description
namelist	See above.

---

Note: The Media Content Recovery extension currently supports audio content only.

---

**<disconnect>**

Causes the interpreter context to disconnect from the user. As a result, the interpreter context generates an event (`telephone.disconnected.hangup`). Upon receiving the event, the application then performs a cleanup.

`<disconnect>` is the only supported VoiceXML call control tag (the same effect as `<exit>`).

The value of this tag must be empty.

**Parents:** `<block>`, `<catch>`, `<error>`, `<filled>`, `<help>`, `<if>`, `<noinput>`, `<nomatch>`

**Children:** None

**Attributes:** None.

**<else>**

Marks the beginning of content to be executed when the parent `<if>` tag and all `<elseif>` tags at the same level of nesting have conditions that evaluate to false. Ends at the closing `<if>`.

The value of this tag must be empty.

**Parent:** `<if>`

**Attributes:** None.

**<elseif>**

Marks the beginning of content to be executed when the parent `<if>` and all `<elseif>` tags evaluate to false and the `cond` attribute evaluates to true. Ends at the next `<elseif/>` or `<else>` or closing `<if>` tag, whichever comes first.

The value of this tag must be empty.

**Parent:** `<if>`

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.

**<enumerate>**

The <enumerate> element is an automatically generated description of the choices available to the user. It specifies a template that is applied to each choice in the order they appear in the <menu> element, or in the <field> element that contains <option> elements.

Note that the <enumerate> element may only be used within the prompts and the catch elements associated with <menu> elements and with <field> elements that contain <option> elements.

**Parent:** <audio>, <catch>, <error>, <field>, <help>, <if>, <menu>, <noinput>, <nomatch>, <prompt>

**Children:** <audio>, <value>

**Attributes:** None.

**<error>**

This element receives (catches) an error event. It is shorthand for <catch event="error"> and receives all events of type 'error'.

The value can be empty or can contain one or more children listed below where the element is either parsed text or the tag.

**Parents:** <field>, <form>, <initial>, <menu>, <record>, <subdialog>, <transfer>, <vxml>

**Children:** <assign>, <audio>, <clear>, <disconnect>, <enumerate>, <exit>, <goto>, <if>, <prompt>, <reprompt>, <return>, <script>, <submit>, <throw>, <value>, <var>

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
count	The occurrence of the event. (Default is 1.) The count allows you to handle different occurrences of the same event differently. Each form item and <menu> maintains a counter for each event that occurs while it is being visited; these counters are reset each time the <menu> or form item's <form> is re-entered.

**<exit>**

Halts all loaded documents and returns control to the interpreter context. Once <exit> returns control to the interpreter context, the interpreter context is free to do as it wishes. For example, it can play a top level menu for the user, drop the call, or the user to an operator.

The value of this tag must be empty.

**Parents:** [<block>](#), [<catch>](#), [<error>](#), [<filled>](#), [<help>](#), [<if>](#), [<noinput>](#), [<nomatch>](#)

**Attributes:** The IP Media Server returns no values and ignores the values specified in the `expr` or `namelist` attributes if they are supplied.

### **<field>**

An input field that collects prompt-solicited user input within a form. Acceptable input is specified by `type`.

The value can be empty or can contain one or more of the children listed below.

**Parent:** [<form>](#)

**Children:** [<audio>](#), [<catch>](#), [<enumerate>](#), [<error>](#), [<filled>](#), [<help>](#), [<if>](#), [<link>](#), [<noinput>](#), [<nomatch>](#), [<prompt>](#), [<property>](#), [<value>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
modal	If FALSE (default), all active grammars are turned on while collecting this field. If TRUE, only the field's grammars are enabled: all others are temporarily disabled.
name	Field-item variable in the dialog scope that holds the result.
slot	Name of the grammar slot used to populate the variable. If it is absent, it defaults to the variable name. Name of the grammar slot used to populate the variable. If the grammar slot is absent, it defaults to the variable name. This attribute is useful when the grammar format being used has a mechanism for returning sets of slot/value pairs and the slot names differ from the field item variable names. If the grammar returns only one slot, as do the built-in type grammars like Boolean, then no matter what the slot's name, the field item variable gets the value of that slot.
type	Type of field, for example digits, currency, phone, date, Boolean, number, time. If not present, <grammar> and/or <dtmf> elements can be specified instead.

Note: An error event may be thrown if a reserved ECMAScript word is used as the name. If the type attribute references an invalid builtin type, the `error.unsupported.builtin` event is thrown at runtime.

**<filled>**

Specifies an action to perform when some combination of fields are filled by user input.

The value can be empty or can contain one or more of the children listed below.

**Parents:**     <field>, <form>, <record>, <subdialog>, <transfer>

**Children:**   <assign>, <audio>, <clear>, <disconnect>, <exit>, <goto>, <if>, <log>, <prompt>, <reprompt>, <return>, <script>, <submit>, <throw>, <value>, <var>



**Attributes:**

Attribute	Description
mode	Any or all (default). If any, action is executed when any field is filled by last user input, If all, action executes when all fields are filled.
namelist	The input items on which to trigger. It defaults to the names of the form's field items.

Note: An error.badfetch event is thrown if mode or namelist is specified for a <filled> within an input item (ex. in a <field>).

**<font>**

Specifies the font used in a text overlay. For an example of text overlay, see [“Text Overlay” \(page 304\)](#). For more on text overlay, see [“Text Overlay Functionality” \(page 236\)](#).

**Parents:** [<content>](#)

**Children:** none

**Attributes:**

Attribute	Description
name	Name of the font. Specify the URL to the font file using this format: font:///usr/share/fonts/... Or for cached fonts, specify the name defined in trcr.cfg. The default value is font:///usr/share/fonts/liberation/LiberationSans-Regular.ttf.
size	Font size. The default value is 14.
style	Used by cached fonts only, which are defined in trcr.cfg. For dynamic fonts, the style is defined in the font file. One or more attributes to apply to the text. Values are: normal, bold, italic, underlined, and outlined. The default value is normal.

Attribute	Description
fgcolor	<p>Color of text. Values are: red, blue, green, yellow, magenta, cyan, white, black, transparent or an unsigned integer.</p> <p>If a color enumeration is specified, the color will be opaque.</p> <p>If an integer is specified, use the format 0xRRGGBBTT, where RR is a 2-digit hexadecimal value for the red component, GG is for the green component, BB is for the blue component, and TT is the value for the transparency component.</p> <p>Transparent can be used as a color or specified in the TT of the bitmask.</p> <p>The default value is white.</p> <p>For background color of text, see <a href="#">&lt;overlay&gt;</a> element.</p>

### <form>

One of two kinds of dialogs for collecting user input. The other is <menu>.

The value of this tag can be empty or can contain one or more of the child tags listed below.

**Parent:** [<vxml>](#)

**Children:** [<block>](#), [<catch>](#), [<error>](#), [<field>](#), [<filled>](#), [<grammar>](#), [<help>](#), [<initial>](#), [<link>](#), [<noinput>](#), [<nomatch>](#), [<object>](#), [<property>](#), [<record>](#), [<subdialog>](#), [<var>](#)

**Attributes:**

Attribute	Description
id	Name of the form.
scope	The scope of the form's grammar. Defaults to the dialog scope.

Note: An error.badfetch event is thrown if multiple forms have the same id (on the same page).

### <goto>

Used in executable content to cause a transition to another form item in the current form, another dialog in the current document, or another document.

The value of this tag must be empty.

**Parents:** [<block>](#), [<catch>](#), [<error>](#), [<filled>](#), [<help>](#), [<if>](#), [<noinput>](#), [<nomatch>](#)

**Attributes:** One of the following values must be specified:

Attribute	Description
expr	ECMAScript expression that yields the target URI.
expriitem	ECMAScript expression that yields the name of the target form item.
next	URL to which to transition.
nextitem	Name of the next form item in the current form.
fetchaudio	The URI of audio to play while waiting for the document to be fetched. Optional.
fetchtimeout	The length of time to wait for the next document to be fetched before throwing an error.badfetch event. Optional.

### <grammar>

Defines a grammar for speech or DTMF recognition. The IP Media Server supports the XML Form of the W3C Speech Recognition Grammar Specification (SRGS).

**Parents:**     [<choice>](#), [<field>](#), [<form>](#), [<link>](#), [<record>](#), [<transfer>](#)

## Attributes:

Attribute	Description
xml:lang	The language identifier of the grammar. The list of supported values for this attribute depends on the ASR engine being used. Optional.
src	The URI specifying the location of the grammar file. The URI can be one of the following formats: <ul style="list-style-type: none"> <li>♦ External grammar file: URL of the grammar file</li> <li>♦ Built-in grammars: builtin:grammar/type (refer to &lt;field&gt; element for built-in types)</li> </ul>
scope	The scope of the grammar. This attribute can be defined only if the <grammar> element is the child of a <form> or <menu> element. Optional. (Defaults to dialog.) <ul style="list-style-type: none"> <li>♦ dialog - This grammar is only active within the form.</li> <li>♦ document - This grammar is active throughout the document. If the current page is an application root document, then the grammar is active throughout the application.</li> </ul>
type	The MIME type of the grammar format. The IP Media Server supports the "application/srgs+xml " MIME type.
mode	The mode of the grammar. Optional. (Defaults to voice.) <ul style="list-style-type: none"> <li>♦ voice - voice input</li> <li>♦ dtmf - DTMF input (replaces the &lt;dtmf&gt; element)</li> </ul>
root	The root rule of the grammar. The default value depends on the ASR engine being used. Optional.
version	The version of the grammar. Optional. (Defaults to 1.0.)
weight	Specifies the weight of the grammar.
tag-format	The format that will be used to define tags, for slot filling. Support for this attribute depends on the ASR engine being used.
xml:base	Declares the base URI from which relative URIs in the grammar are resolved. This base declaration has precedence over the <vxml> base URI declaration. Optional.
fetchhint	Defines when the grammar file should be fetched. Optional. <ul style="list-style-type: none"> <li>♦ prefetch - grammar file may be downloaded when the page is loaded</li> <li>♦ safe - only load the grammar file when needed</li> </ul>
fetchtimeout	The length of time to wait for the grammar file to be fetched before throwing an error.badfetch event. Optional.

Attribute	Description
maxage	Indicates that this document is willing to use a cached copy of the grammar file only while the age of the cached copy is less than or equal to the number of seconds specified by this attribute. Optional.
maxstale	Indicates that this document is willing to use a cached copy of the grammar file that has exceeded its expiration time by as much as the number of seconds specified by this attribute. Optional.

**Note:** Referencing an unsupported built-in grammar will result in an error.unsupported.builtin being thrown.

Speech grammars require an ASR resource to be configured and available or else an error.noresource event will be thrown.

DTMF grammars are handled local to the IP Media Server and do not required an ASR resource. Refer to [DTMF Grammars Guidelines on page 187](#) for more information.

## <help>

Receives a help event, and is shorthand for <catch event="help">. The value can be empty or can contain one or more of the children listed below.

**Parents:** <field>, <form>, <initial>, <menu>, <record>, <subdialog>, <transfer>, <vxml>

**Children:** <assign>, <audio>, <clear>, <disconnect>, <exit>, <goto>, <if>, <log>, <prompt>, <reprompt>, <return>, <script>, <submit>, <throw>, <value>, <var>

**Attributes:**

Attribute	Description
count	The occurrence of the event. (Default is 1.) The count allows you to handle different occurrences of the same event differently. Each form item and <menu> maintains a counter for each event that occurs while it is being visited; these counters are reset each time the <menu> or form item's <form> is re-entered.
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.

## <if>

Used for conditional logic. It has optional <else> and <elseif> elements.

The value can be empty or can contain one or more of the children listed below.

**Parents:** [<block>](#), [<catch>](#), [<error>](#), [<filled>](#), [<help>](#), [<if>](#), [<noinput>](#), [<nomatch>](#)

**Children:** [<assign>](#), [<audio>](#), [<clear>](#), [<disconnect>](#), [<else>](#), [<elseif>](#), [<exit>](#), [<goto>](#), [<if>](#), [<log>](#), [<prompt>](#), [<reprompt>](#), [<return>](#), [<script>](#), [<submit>](#), [<throw>](#), [<value>](#), [<var>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.

### <initial>

Declares initial logic upon entry into a mixed-initiative form.

The value can be empty or can contain one or more of the children listed below.

**Parent:** [<form>](#)

**Children:** [<audio>](#), [<catch>](#), [<error>](#), [<help>](#), [<link>](#), [<noinput>](#), [<nomatch>](#), [<prompt>](#), [<property>](#), [<value>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
name	The name of a form-item variable used to track whether the <initial> is eligible to execute; defaults to an inaccessible internal variable.

---

Note: An error event may be thrown if a reserved ECMAScript word is used as the name.

---

### <link>

Specifies a destination for a transition or an event to be thrown when input matches.

The value of this tag can be empty or can contain one or more grammar tags.

**Parents:** [<field>](#), [<form>](#), [<initial>](#), [<vxml>](#)

**Children:** [<grammar>](#)

**Attributes:** Must specify one of next, expr, or event.

Attribute	Description
next	URI to go to. Either a document (perhaps with an anchor to specify the starting dialog), or a dialog in the current document (just a bare anchor).
expr	Like next, except that the URI is dynamically determined by evaluating the given ECMAScript expression.
event	The event to generate when the user matches one of the link grammars.
eventexpr	An ECMAScript expression to be evaluated and used as the name of the event to be thrown when the user input matches one of the link grammars. Exactly one of next, expr, event, or eventexpr must be specified.
fetchtimeout	Interval to wait for the content to be returned before generating an error.badfetch event. If not specified, a value derived from the innermost fetchtimeout property is used.
dtmf	The DTMF sequence for this link. It is equivalent to a simple DTMF grammar. This attribute can be used at the same time as other grammars: the link is activated when user input matches either a link grammar or this DTMF sequence. Optional.
fetchaudio	The URI of audio to play while waiting for the document to be fetched. Optional.
maxage	Indicates that this document is willing to use a cached copy of the next document only while the age of the cached copy is less than or equal to the number of seconds specified by this attribute. Optional.
maxstale	Indicates that this document is willing to use a cached copy of the next document that has exceeded its expiration time by as much as the number of seconds specified by this attribute. Optional.
message	A message string providing additional context about the event being thrown. The message will be available as a variable within the scope of the <catch> element. Optional. (Can only be used if event or eventexpr is specified; only one of message and messageexpr can be specified.)
messageexpr	An ECMAScript expression to be evaluated and used as a message string, as documented under message, above. Optional. (Can only be used if event or eventexpr is specified; only one of message and messageexpr can be specified.)

**Note:** The event specified by event/eventexpr is thrown based on the scope of the active dialog (ie. <field>, <menu>, etc., where the user was when the link grammar was

matched). All attribute values that are ECMAScript expressions are evaluated based on variables from the scope of the active dialog (ie. <field>, <menu>, etc., where the user was when the link grammar was matched). An error.badfetch event is thrown if more or less than one of next, expr, event, and eventexpr are specified.

---

### <log>

The <log> element allows the application to generate messages for debugging.

**Parents:**     <log>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Children:**   <value>

**Attributes:**

Attribute	Description
dest	When set to SNMP, this instructs the browser to send out the msVXMLCriticalError trap to all SNMP trap hosts configured on the IP Media Server.  The text value of the <log> element can be used to contain the details about the error. This content is placed in the msVXMLLastCriticalError object. This object is passed as a varbind of the msVXMLCriticalError trap. (Default: none.)  Following is an example <log> tag with the trap extension: <code>&lt;log dest="snmp"&gt;Unable to contact VoiceXML application server.&lt;/log&gt;</code>
expr	An ECMAScript expression of which the evaluated value will be appended to the <log> element content and label attribute value. Optional.
label	A string label that will be appended to the <log> element content before the value of the expr attribute is appended. Optional.

### <menu>

Dialog for making a selection among choices.

The value can be empty or can contain one or more of the children listed below.

**Parents:**     <vxml>

**Children:**   <audio>, <catch>, <choice>, <enumerate>, <error>, <help>, <noinput>, <nomatch>, <prompt>, <property>, <value>



**Attributes:**

Attributes	Description
id	Identifier of the menu. It allows the menu to be the target of a <goto> or a <submit>.
scope	Menu's grammar scope. Either dialog (default) or document.
dtmf	TRUE or FALSE. When TRUE, any choices that do not have explicit DTMF elements are given implicit ones: 1, 2, etc. When FALSE, touch-tone digits not assigned.
accept	Specifies how to generate the default grammar for each <choice> element. Optional. (Defaults to exact.) <ul style="list-style-type: none"> <li>exact - The text contained in the &lt;choice&gt; element defines the exact phrase to be recognized.</li> <li>approximate - The text contained in the &lt;choice&gt; element defines an approximate recognition phrase. A subset of the words in the phrase expression can be matched. For example, "hello world" can be matched with "hello world", "hello", or "world".</li> </ul>

**<meta>**

Page properties as meta-data, as in HTML.

---

Note: This is ignored by the VoiceXML Interpreter.

---

The value of this tag must be empty.

**Parent:**     <vxml>

**Attributes:**

Attribute	Description
name	Name of the meta-data property.
http-equiv	Name of an HTTP response header. Either name or http-equiv must be specified, but not both.
content	Value of the meta-data property. Required.

**<noinput>**

A type of catch element event used when the user does not respond within the required timeout interval. For example, <noinput> Shorthand for <catch event="noinput">. See <catch>.

The value can be empty or can contain one or more of the children listed below.

**Parents:** <field>, <form>, <initial>, <menu>, <record>, <subdialog>, <transfer>, <vxml>

**Children:** <assign>, <audio>, <clear>, <disconnect>, <enumerate>, <exit>, <goto>, <if>, <prompt>, <reprompt>, <return>, <script>, <submit>, <throw>, <value>, <var>

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
count	A number that allows you to issue different prompts if the user is doing something repeatedly. If omitted, it defaults to 1.

### <nomatch>

Used to catch a nomatch event. Shorthand for <catch event="nomatch">. See “<catch>” (page 192).

The value can be empty or can contain one or more of the children listed below.

**Parents:** <field>, <form>, <initial>, <menu>, <object>, <record>, <subdialog>, <transfer>, <vxml>

**Children:** <assign>, <audio>, <clear>, <disconnect>, <enumerate>, <exit>, <goto>, <if>, <log>, <prompt>, <reprompt>, <return>, <script>, <submit>, <throw>, <value>, <var>

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
count	A number that allows you to issue different prompts if the user is doing something repeatedly. If omitted, it defaults to 1.

### <object>

No IP Media Server platform-specific objects are currently defined. Therefore, any use of <object> will result in the following error thrown:

```
error.unsupported.objectname
```

Objectname is a fixed string; it is not substituted with the name of the object.

**<option>**

Specifies an option in a <field>. It is a convenient way to specify a list of choices in a field without specifying a grammar. The grammar is generated automatically from the text contained in each <option> and the optionally specified dtmf sequence in each <option>.

**Parent:**     <field>

**Attributes:**

Attribute	Description
dtmf	A DTMF sequence to select this option. Optional.
accept	Specifies how to generate the grammar for this <option> element. Optional. (Defaults to exact.) <ul style="list-style-type: none"> <li>exact - The text contained in the &lt;option&gt; element defines the exact phrase to be recognized.</li> <li>approximate - The text contained in the &lt;option&gt; element defines an approximate recognition phrase. A subset of the words in the phrase can be matched. For example, "hello world" can be matched with "hello world", "hello", or "world".</li> </ul>
value	The string to assign to the field variable when this option is selected, either by speech or DTMF. Optional. (Defaults to the option text with leading and trailing white space removed, if specified. Otherwise, defaults to the value of the dtmf attribute.)

**<overlay>**

Specifies a text overlay for a video stream. For an example of text overlay, see “Text Overlay” (page 304). For more on text overlay, see “Text Overlay Functionality” (page 236).

**Parent:**     <prompt>

**Children:**   <content>

**Attributes:**

Attribute	Description
position X<int>Y<int>	The X coordinate and Y coordinate of the upper left-hand corner of the overlay area, given either in pixels or as a percentage of the background video frame. To specify a percentage, include % character in the value. The X and Y values can independently use pixels or a percentage value. Example: <overlay position=x20y50%>.

Attribute	Description
size X<int>Y<int>	<p>The height (X coordinate) and width (Y coordinate) of the overlay area, given either in pixels or as a percentage of the background video frame.</p> <p>To specify a percentage, include % character in the value.</p> <p>The X and Y values can independently use pixels or a percentage value.</p> <p>Example: &lt;overlay size=x45%y20&gt;.</p>
delay	<p>Delays the start of the overlay until a specified number of seconds has passed from the start of the video stream.</p> <p>Default value is 0.</p>
layer	<p>The order in which overlays are displayed when multiple overlays are specified.</p> <p>An overlay with a higher number (such as 5) is displayed in the foreground relative to an overlay with a lower number (such as 4).</p> <p>When using multiple overlays, be careful that the overlays do not overlap and render the text unreadable. By default, the same layer value is used for all overlays.</p> <p>Default value is 0.</p>
border-width	<p>Border width of the overlay area given in pixels.</p> <p>Default value is 0.</p>
border-color	<p>Border color of the overlay area. Values are: red, blue, green, yellow, magenta, cyan, white, black, transparent or an unsigned integer.</p> <p>If a color enumeration is used, the color will be opaque.</p> <p>If an integer is specified, use the format 0xRRGGBBTT, where RR is a 2-digit hexadecimal value for the red component, GG is for the green component, BB is for the blue component, and TT is the value for the transparency component.</p> <p>Transparent can be used as a color or specified in the TT of the bitmask.</p> <p>Default value is black.</p>
timeout	<p>Stops displaying the overlay after a specified number of seconds has passed from the start of the video stream.</p> <p>A value of 0 means that no timeout will occur.</p> <p>Default value is 0.</p>

Attribute	Description
bgcolor	<p>Background color of the text (the space between the letters). Values are: red, blue, green, yellow, magenta, cyan, white, black, transparent or an unsigned integer.</p> <p>If a color enumeration is used, the color will be opaque.</p> <p>If an integer is specified, use the format 0xRRGGBBTT, where RR is a 2-digit hexadecimal value for the red component, GG is for the green component, BB is for the blue component, and TT is the value for the transparency component.</p> <p>Transparent can be used as a color or specified in the TT of the bitmask.</p> <p>Default value is transparent.</p> <p>For foreground color of the text, see <a href="#">&lt;font&gt;</a>.</p>

**<param>**

Specifies the values that are passed to subdialogs.

The value of this tag must be empty.

**Parent:**     [<object>](#), [<subdialog>](#)

**Attributes:**

Attribute	Description
name	Name to be associated with this parameter when the subdialog is invoked. Required.
expr	Expression that computes the value associated with name.
value	Associates a literal string value with name.
valuetype	Whether the value associated with name is data or a URI (ref). Default is data.
type	MIME type of the result provided by a URI if the value type is ref. Only relevant for uses of <a href="#">&lt;param&gt;</a> in <a href="#">&lt;object&gt;</a> .

**<prompt>**

Controls the output of pre-recorded audio, video, multimedia or TTS to the user.

The value can be empty or can contain one or more of the children listed below, where the element is either parsed text or the tag.

**Parents:**     [<block>](#), [<catch>](#), [<error>](#), [<field>](#), [<filled>](#), [<help>](#), [<if>](#),  
[<initial>](#), [<menu>](#), [<noinput>](#), [<nomatch>](#), [<object>](#) [<return>](#),  
[<subdialog>](#), [<transfer>](#)

**Children:** [<audio>](#), [<enumerate>](#), [<overlay>](#), [<ssml>](#) [<value>](#)

**Attributes:**

Attribute	Description
bargein	Whether a prompt may be interrupted. True - Allow input to interrupt (Default) False - Do not allow input to interrupt
bargeintype	The bargein type. Optional. (Defaults to speech.) <ul style="list-style-type: none"> <li>speech - Any user utterance can barge in on the prompt.</li> <li>hotword - Only user input that matches a grammar can barge in on the prompt. (Not supported in the current release.)</li> </ul> Overrides the value of the bargeintype property.
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
count	A number that allows you to issue different prompts if the user is doing something repeatedly. If omitted, it defaults to 1.
timeout	How long to wait for the next input. The default noinput timeout is 5 seconds.
vcr	Whether VCR controls are to be active for the prompt block. (Default: FALSE.)
xml:lang	The language identifier of the prompt. If omitted, it defaults to the value specified in the document's xml:lang attribute. Optional.  The list of supported values for this attribute depends on the TTS engine being used. The xml:lang must always match the TTS engine language. Refer to your vendor documentation for support details.
xml:base	Declares the base URI from which relative URIs in the prompt are resolved. This base declaration has precedence over the <vxml> base URI declaration. If a local declaration is omitted, the value is inherited down the document hierarchy. Optional.

### <property>

Sets a property value that affects platform behavior.

The value of this tag must be empty.

**Parents:** [<field>](#), [<form>](#), [<initial>](#), [<menu>](#), [<object>](#), [<record>](#), [<subdialog>](#), [<transfer>](#), [<vxml>](#)

**Attributes:**

Attribute	Description
name	Property name. Required.
value	Property value. Required.

**<record>**

A field item that collects a recording from the user. The recording is stored in the field item variable, which can be played back or submitted to the server. Only external storage is available.

The value can be empty or can contain one or more of the children listed below.

**Parent:**        [<form>](#)

**Children:**    [<audio>](#), [<catch>](#), [<error>](#), [<filled>](#), [<grammar>](#), [<help>](#),  
[<noinput>](#), [<nomatch>](#), [<prompt>](#), [<property>](#), [<value>](#)

**Attributes:**

Attribute	Description
beep	TRUE or FALSE (the default). If TRUE, the application plays a tone just prior to recording.
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
dtmfterm	TRUE or FALSE. If TRUE (default), a DTMF keypress terminates the recording. If FALSE, a DTMF keypress does not terminate recording.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
finalsilence	Interval of silence that indicates the end of speech. (Default: 3 seconds.)
meantime	If the duration of the recording is less than this attribute, then the recording is assumed to be empty and a nonbook is thrown. Minimum value is 250ms (if defined as a shorter duration, it will be set to 250ms); maximum value is the value of the maxtime attribute (if defined as a longer duration, it will be set to 250ms). Optional. (Defaults to 250ms.)
maxtime	Maximum duration to record. (Default: 10 seconds.)
name	The name of a form-item variable used to track whether the <initial> is eligible to execute; defaults to an inaccessible internal variable.
type	MIME type of the recording.

Note: Voice and DTMF grammars are not active during the record. DTMF digits can only be used to terminate the recording when the record elements 'dtmfterm' attribute is set to true.

---

### <reprompt>

Used inside executable content to set a flag indicating that a new attempt should be made to issue a prompt for the current item.

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attributes:** None.

### <return>

Returns execution of a subcollege and returns control and data to the calling dialog.

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attributes:**

Attribute	Description
event	Returns, then sends this event.
namelist	Variable names to be returned to the calling dialog.
eventexpr	An ECMAScript expression to be evaluated and used as the name of an event to be returned to the calling dialog and thrown. Exactly one of event, eventexpr, or namelist may be specified.
message	A message string providing additional context about the event being thrown. The message will be available as a variable within the scope of the <catch> element. Optional. (Can only be used if event or eventexpr is specified; only one of message or messageexpr can be specified.)
messageexpr	An ECMAScript expression to be evaluated and used as the message string, as documented under message, above. Optional. (Can only be used if event or eventexpr is specified; only one of message or messageexpr can be specified.)



**<say-as>**

Type of text construct contained within the element that defines the way a word or phrase is spoken, including text generated by the <value> tag. The rules to convert <sayas> text into pre-recorded speech fragments are determined by an external script. The default URI of this script is <http://localhost/snowshore/vxmlphrase.cgi>. This URI can be modified to use an external server by the `com.snowshore.recsrc` property.

The default rules handle US English only. The value of this tag must be parsed text.

---

**Note:** The <say-as> element is supported with IP Media Server resources and not a TTS server. It provides an alternative to TTS for minimal phrases.

---

**Parents:**     [<audio>](#), [<choice>](#), [<emphasis>](#), [<enumerate>](#), [<paragraph>](#),  
                   [<p>](#), [<prompt>](#), [<prosody>](#), [<sentence>](#), [<s>](#), [<speaking>](#), [<voice>](#)

**Children:** [<value>](#)

**Attributes:**

Attribute	Description
interpret-as	Describes the way to render the value of: <ul style="list-style-type: none"> <li>• phone (NANP numbers)</li> <li>• digits (Each spoken with no pauses.)</li> <li>• number (Spoken as a positive cardinal number. Negative numbers are not supported and will produce odd output.)</li> </ul>

**<script>**

Allows the specification of a block of client-side ECMAScript code.

---

**Note:** Although the Dialogic® IP Media Server supports this tag, Dialogic does not recommend its use and considers it inadvisable to execute arbitrary logic on the IP Media Server due to resource limitations.

---

The value of this tag must be parsed text.

**Parents:**     [<block>](#), [<catch>](#), [<error>](#), [<filled>](#),[<form>](#), [<help>](#), [<if>](#),  
                   [<noinput>](#), [<nomatch>](#), [<vxml>](#)

**Attributes:**

Attribute	Description
src	URL for the resource.
chariest	The character encoding if an external script is used. Optional.

Attribute	Description
fetchhint	Defines when the script should be fetched. Optional. <ul style="list-style-type: none"> <li>♦ perfect - script may be downloaded when the page is loaded</li> <li>♦ safe - only load the script when needed</li> </ul>
fetchtimeout	The length of time to wait for the script to be fetched before throwing an error.badfetch event. Optional.
maxage	Indicates that this document is willing to use a cached copy of the script only while the age of the cached copy is less than or equal to the number of seconds specified by this attribute. Optional.
maxstale	Indicates that this document is willing to use a cached copy of the script that has exceeded its expiration time by as much as the number of seconds specified by this attribute. Optional.

### <scroll>

Specifies the characteristics of scrolling text in a text overlay. This is an optional element for a text overlay. If the <scroll> element is not used, the text overlay will be static. For an example of text overlay, see [“Text Overlay” \(page 304\)](#). For more on text overlay, see [“Text Overlay Functionality” \(page 236\)](#).

**Parents:**     <content>

**Children:**   none

**Attributes:**

Attribute	Description
mode	The mode of the scrolling text. Values are: content (scroll one time), continuous (scroll continuously), and none. Required attribute. Default is none.
speed	Speed of scrolling measured in pixels per second. Default is 25.
direction	Direction of scrolling. Values are: left, right, up, down. Default is left.
term-delay	Used with content mode. Termination delay. The amount of time in seconds to keep the overlay in the video after scrolling completes. Default is 5 seconds.

**<subdialog>**

Invokes a second dialog within the current dialog. It provides a way for invoking a new interaction, and returning to the original form. Local data, grammars, and state information are saved and available when returning to the calling document.

The value can be empty or can contain one or more of the children listed below.

**Parent:**        [<form>](#)

**Children:**    [<audio>](#), [<catch>](#), [<error>](#), [<filled>](#), [<help>](#), [<noinput>](#),  
[<nomatch>](#), [<param>](#), [<prompt>](#), [<property>](#), [<value>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped. the form is not visited.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
method	Request method (get or post).
modal	Controls which grammars are active during the subcollege.
name	The result returned from the subdialog, an ECMAScript object whose properties are the ones defined in the namelist attribute of the <return> element.
src	URI of the subdialog. Required.
srcexpr	An ECMAScript expression to be evaluated and used as the URI of the subdialog. This can be the URI of another document, a dialog in the same document, or a dialog in another document. If this attribute is empty (""), or set to the empty string (""), the current document will be used. Exactly one of src or srcexpr must be specified.
namelist	A space-separated list of variables to submit to the subdialog. Optional. (Defaults to nothing.)
method	The request method: get or post. Optional. (Defaults to get.)
enctype	The MIME encoding of the submitted data. The following types are supported: <ul style="list-style-type: none"> <li>• application/x-www-form-urlencoded</li> <li>• multipart/form-data</li> </ul> Optional. (Defaults to application/x-www-form-urlencoded.)
fetchaudio	The URI of audio to play while waiting for the document to be fetched. Optional

Attribute	Description
fetchhint	Defines when the subdialog should be fetched. Optional. safe - only load the subdialog when needed
fetchtimeout	The length of time to wait for the subdialog to be fetched before throwing an error.badfetch event. Optional.
maxage	Indicates that this document is willing to use a cached copy of the subdialog only while the age of the cached copy is less than or equal to the number of seconds specified by this attribute. Optional.
maxstale	Indicates that this document is willing to use a cached copy of the subdialog that has exceeded its expiration time by as much as the number of seconds specified by this attribute. Optional.

### <submit>

This element is like <goto> because the application gets a new document. Unlike <goto>, it lets you submit a list of variables to the document server, using an HTTP GET or POST request.

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

### Attributes:

Attribute	Description
expr	Like next, except that the URI is dynamically determined by evaluating the given ECMAScript expression. One of next, expr, or fetchtimeout is required.
method	The request method (GET or POST). The default is GET.
namelist	The list of variables to submit.
next	The URL to which the query is submitted.
fetchtimeout	Interval to wait for the content to be returned before generating an error.badfetch event. If not specified, a value derived from the innermost fetchtimeout property is used.
enctype	The MIME encoding of the submitted data. The following types are supported: <ul style="list-style-type: none"> <li>◆ application/x-www-form-urlencoded</li> <li>◆ multipart/form-data</li> </ul> Optional. (Defaults to application/x-www-form-urlencoded.)

Attribute	Description
fetchaudio	The URI of audio to play while waiting for the document to be fetched. Optional
fetchhint	Defines when the subdialog should be fetched. Optional. <ul style="list-style-type: none"> <li>♦ safe - only load the subdialog when needed</li> </ul>
maxage	Indicates that this document is willing to use a cached copy of the subdialog only while the age of the cached copy is less than or equal to the number of seconds specified by this attribute. Optional.
maxstale	Indicates that this document is willing to use a cached copy of the subdialog that has exceeded its expiration time by as much as the number of seconds specified by this attribute. Optional.

---

**Note:** An error.semantic event is thrown if an undeclared variable name is used in the namelist. An error.badfetch event is thrown if more or less than one of next and expr are specified.

---

### **<throw>**

Sends an event to be received by <catch>.

The value of this tag must be empty.

**Parents:**    <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attribute:**

Attribute	Description
event	Returns, then sends this event.
eventexpr	An ECMAScript expression to be evaluated and used as the name of the event to be thrown. Exactly one of event or eventexpr must be specified.
message	A message string providing additional context about the event being thrown. For the pre-defined events thrown by the platform, the value of the message is platform-dependent. The message will be available as a variable within the scope of the <catch> element. Optional. (Only one of message or messageexpr can be specified.)
messageexpr	An ECMAScript expression to be evaluated and used as a message string, as documented under message, above. Optional. (Only one of message or messageexpr can be specified.)

---

**Note:** An error.badfetch event is thrown if more or less than one of event and eventexpr are specified, or if more than one of message and messageexpr are specified.

---

**<transfer>**

Provides the ability to place an outgoing voice call to support transfers. A “bridged” transfer session returns the caller to the original session with the interpreter upon completion. A “blind” transfer session terminates when completed.

**Parent:**     [<form>](#)

**Attributes:**

Attribute	Description
name	<p>The outcome of the transfer attempt. For possible return values, see the following table.</p> <p>Note: If the transfer request is unacceptable and the transfer is not initiated at all, the transfer variable will not be set and the platform will throw an error, instead.</p>
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
dest	The URL of the destination (phone, IP telephone address). (Default: none)
destexpr	<p>An ECMAScript expression yielding the URI of the destination.</p> <p>The expression must produce a valid SIP URI, for example sip:19783678400@gateway.carrier.net. (Default: none.)</p>
bridge	<p>Determines what to do once the call is connected.</p> <p>A value of 'true' indicates the transfer is bridged.</p> <p>A value of 'false' indicates a blind transfer.</p> <p>A blind transfer is not currently supported by the IP Media Server. A value of false will result in the error.unsupported.transfer.blind event being thrown. A bridged transfer is currently implemented by hairpinning media from both calls through the IP Media Server, which requires the VoiceXML session to continue its existence for the lifetime of the call transfer.</p>
src	The SIP URI that shows up as ANI at the callee. Must be of the form sip:user@server. This is to identify the caller for billing purposes. (Dialogic-specific attribute)
srcexpr	An ECMAScript expression yielding src. (Dialogic-specific attribute).
stopdigits	Defines the list of single DTMF digits that ends the transfer.
longdigit	<p>Enables detection of long DTMF digits or multiple, short instance of the same digit.</p> <p>Indicates that the transfer can be terminated only with a long DTMF digit or multiple, short instances of the same digit.</p> <p>YES turns DTMF detection on for long digits. NO (default) turns DTMF detection off for long digits.</p>
type	Specifies the transfer behavior (from the VoiceXML application's perspective). Optional.

Attribute	Description
video	Whether video is available on the outbound call. The value can be one of suppress or allow. See <a href="#">“RTP Codec Selection with VoiceXML &lt;transfer/&gt;”</a> (page 378).
connecttimeout	The time to wait while trying to connect the call before returning, and setting the transfer variable to 'noanswer'. Minimum value allowed is 5s. This attribute only applies for bridge or consultation transfers. Optional. (Defaults to 30s.)
maxtime	The maximum time that the call is allowed to last. Minimum value is 30 seconds; maximum is one week. A value of 0 represents no limit. This attribute only applies for bridge transfers. Optional. (Defaults to 0.)
transferaudio	The URI of an audio file to be played while connecting the call. Transfer audio cannot be played if connectwhen="immediate"; in this case, network ringing will be heard instead. Optional.
aai	A string containing Application-to-Application Information (AAI) data to be sent to an application on the far-end.
aaiexpr	An ECMAScript expression evaluating to the Application-to-Application Information (AAI) data string, to be sent to an application on the far-end, as documented under aai, above.
requir	The SIP URI to be used as the Request-URI in an outbound INVITE. The value must be compliant with the grammar for a SIP URI. If this attribute is not present, the Request-URI will have the same value as the To header. The To URI value is determined by the dest attribute. This attribute and the requiexpr attribute enable you to route outbound calls to a proxy. The proxy will modify the request as needed based on the URI parameters placed in the INVITE and relevant network and subscriber data.
requi expr	An ECMAScript expression that evaluates to a SIP URI.

Note: The src, srcexpr, stopdigits, longdigit, and video attributes are Dialogic-specific attributes.

You can use one (but not both) of the following attribute pairs:

- ◆ dest and destexpr
- ◆ requir and requiexpr
- ◆ src and srcexpr
- ◆ aai and aaiexpr



Return values of Name variable:

Value	Description
busy	The endpoint refused the call.
noanswer	There was no answer within the specified time.
network busy	Some intermediate network refused the call.
near_end_disconnect	The call completed and was terminated by the caller.
far_end_disconnect	The call completed and was terminated by the callee.
network_disconnect	The call completed and was terminated by the network.
maxtime_disconnect	The call duration exceeded the value of the maxtime attribute and was terminated by the platform.

**Note:** An error event may be thrown if a reserved ECMAScript word is used as the name. An error.badfetch event is thrown if more or less than one of dest and destexpr are specified, or if more than one of aai and aaiexpr are specified.

## <value>

Inserts the value of an ECMAScript expression into a prompt.

The value of this tag must be empty.

**Parents:** [<audio>](#), [<block>](#), [<catch>](#), [<choice>](#), [<enumerate>](#), [<error>](#), [<field>](#), [<filled>](#), [<help>](#), [<if>](#), [<initial>](#), [<log>](#), [<menu>](#), [<noinput>](#), [<nomatch>](#), [<object>](#), [<prompt>](#), [<record>](#), [<subdialog>](#), [<transfer>](#)

**Attributes:**

Attribute	Description
expr	Expression to render. Required.

## <var>

Declares an ECMAScript variable. The scope of the variable is determined by the parent tag.

The value of this tag must be empty.

**Parents:** [<block>](#), [<catch>](#), [<error>](#), [<filled>](#), [<form>](#), [<help>](#), [<if>](#), [<noinput>](#), [<nomatch>](#), [<vxml>](#)

**Attributes:**

Attribute	Description
name	Name of the variable. Required.
expr	An initial value.

**<vxml>**

The top-level element in a VoiceXML document.

The value of this tag must contain one or more of the children listed below.

**Parent:** None

**Children:** [<catch>](#), [<error>](#) [<form>](#), [<help>](#), [<link>](#), [<menu>](#), [<meta>](#), [<noinput>](#), [<nomatch>](#), [<property>](#), [<script>](#), [<var>](#)

**Attributes:**

Attribute	Description
version	Version of VoiceXML. Set to 2.0. Required.
base	Base URL. The root document URL, if any.
xmlns	The VoiceXML specification requires the use of this attribute (set to <a href="http://www.w3.org/2001/vxml">http://www.w3.org/2001/vxml</a> ) to designate the namespace for VoiceXML.
lang	The language identifier for this document. Used as the language for implicit grammars (i.e., universal grammars, and <code>&lt;choice&gt;/&lt;option&gt;</code> grammars), multi-language Nuance grammars, and <code>&lt;prompt&gt;</code> s. Optional.
application	The URI of this document's application root document. Optional.

---

**Note:** Omitting version or xmlns when they are required will result in an error.badfetch being thrown.

If the root document specified by application cannot be found, an error.badfetch is thrown. If the root document refers to another root document, an error.semantic event is thrown.

---

---

# VoiceXML Properties

## **com.snowshore.criticaldigit\_timer**

Sets a special digit timer that specifies the interval to wait for additional user input before returning a match to an active grammar. This property is used when the application modifies the DTMF matching behavior.

Uses the VoiceXML standard "n {ms} | {s}" timer notation. Default unit is milliseconds (ms).

### **Values:**

<b>Value</b>	<b>Description</b>
-2	The default critical digit timer is set to the same value as the interdigit timer (thus it has no effect).
-1	Immediate. Use "shortest match first".
0	Infinite. Use "longest match first", but waits infinitely once a match is found. Thus, matched input must end with a return digit or restart digit.
>0	Uses "longest match first", and waits that amount of time when a match is found.

## MIME Types Supported

The MIME type and optional codec parameters define the file format and encoding (audio or video) for the file. The MIME type supports both headerless (raw) and .wav file formats. Table 45 lists the supported MIME types, file formats, and audio encodings.

Table 45. Supported MIME Types, Formats, and Audio Encodings

MIME Type	File Format	Audio Encoding	Notes
audio/basic	raw	G.711 $\mu$ -law	
audio/x-alaw-basic	raw	G.711 A law	
audio/vnd.wave	WAVE		IANA standard
audio/wav	WAVE		
audio/x-wav	WAVE		Proprietary
audio/3gpp	3GPP	AMR-NB	
audio/3gpp2	3GPP2	AMR-NB	
video/3gpp	3GPP	AMR-NB	Valid for audio-only file
video/3gpp2	3GPP2	AMR-NB	Valid for audio-only file

There are various commonly-used methods of specifying .wav format and the desired audio encodings. The official IANA registered MIME type for wave is audio/vnd.wave but audio/wav and audio/x-wav are used frequently.

Specify audio encoding by adding a codec parameter and value to the base MIME type, for example:

```
audio/vnd.wave; codec=7
```

The official values for the codec parameter are defined by the IANA registry (see <http://www.iana.org/assignments/wave-avi-codec-registry>), but other commonly used forms are also supported.

Table 46 provides a summary of the supported codec parameters and values and their corresponding audio encodings.

Table 46. Audio Codec Parameter Values

Value	Audio Encoding	Notes
7	G.711 $\mu$ -law	IANA standard
6	G.711 A law	IANA standard
31	Microsoft® GSM	IANA standard

Table 46. Audio Codec Parameter Values

Value	Audio Encoding	Notes
ulaw	G.711 $\mu$ -law	
alaw	G.711 A law	
msgsm	Microsoft® GSM	

Table 47 lists the supported MIME types, file formats, and video encodings.

Table 47. Supported MIME Types, Formats, and Video Encodings

MIME Type	File Format	Encoding	Notes
video/x-wav	WAVE		Proprietary
video/3gpp	3GPP	Audio: AMR-NB. Video: H.263+, H.263++, or H.264.	Supported for play and record.
video/3gpp2	3GPP2	Audio: AMR-NB. Video: H.263+, H.263++, or H.264.	Supported for play only.

The video/x-wav MIME type produces a recording using the same codec as the one negotiated during call setup.

The following code snippet is an example of how to record to a video file using the video/3gpp MIME type. You can replace this type with the preferred MIME type.

```
<record name="vmail_recording" beep="true"
  dtmfterm="true" type="video/3gpp" maxtime="20s"
  ...
  </record>
```

## ECMAScript Functionality

While the media server supports full ECMA (European Computer Manufacturer's Association) Scripting abilities, Dialogic recommends that you perform that type of logic on the application server rather than in the VoiceXML script. This method reduces the processing and memory required by the Dialogic® IP Media Server and allows the IP Media Server to handle a large volume of simultaneous sessions.

Table 48 summarizes the categories and features supported by the IP Media Server's VoiceXML interpreter.

---

**Note:** The ECMAScript Language Specification can be downloaded for free.

---

Table 48. ECMAScript Functionality

Category	Feature/Keyword
Array Handling	Array join, length, reverse, sort
Assignments	Assign (=) Compound Assign (OP=)
Booleans	Boolean
Comments	/*...*/ or //
Constants/Literals	NaN null true, false Infinity undefined
Control flow	Break continue for for...in if...else return while

Table 48. ECMAScript Functionality (continued)

Category	Feature/Keyword
Dates and Time	Date getDate, getDay, getFullYear, getHours, getMilliseconds, getMinutes, getMonth, getSeconds, getTime, getTimezoneOffset, getYear, getUTCDate, getUTCDay, getUTCFullYear, getUTCHours, getUTCMilliseconds, getUTCMinutes, getUTCMonth, getUTCSeconds, setDate, setFullYear, setHours, setMilliseconds, setMinutes, setMonth, setSeconds, setTime, setYear, setUTCDate, setUTCFullYear, setUTCHours, setUTCMilliseconds, setUTCMinutes, setUTCMonth, setUTCSeconds, toGMTString, toLocaleString, toUTCString, parse, UTC
Declarations	Function new this var with
Function Creation	Function arguments, length
Global Methods	Global escape, unescape eval isFinite, isNaN parseInt, parseFloat
Math	Math abs, acos, asin, atan, atan2, ceil, cos, exp, floor, log, max, min, pow, random, round, sin, sqrt, tan, E, LN2, LN10, LOG2E, LOG10E, PI, SQRT1_2, SQRT2
Numbers	Number MAX_VALUE, MIN_VALUE NaN NEGATIVE_INFINITY, POSITIVE_INFINITY
Object Creation	Object new constructor, prototype, instanceof, toString, valueOf

Table 48. ECMAScript Functionality (continued)

Category	Feature/Keyword
Operators	Addition (+), Subtraction (-) Modulus arithmetic (%) Multiplication (*), Division (/) Negation (-) Equality (==), Inequality (!=) Less Than (<), Less Than or Equal To (<=) Greater Than (>) Greater Than or Equal To (>=) Logical And(&&), Or (  ), Not (!) Bitwise And (&), Or ( ), Not (~), Xor (^) Bitwise Left Shift (<<), Shift Right (>>) Unsigned Shift Right (>>>) Conditional (?:) Comma (,) delete, typeof, void Decrement (--), Increment (++)
Objects	Array Boolean Date Function Global Math Number Object String
Strings	String charAt, charCodeAt, fromCharCode indexOf, lastIndexOf split toLowerCase, toUpperCase length



## VoiceXML Extended Session Variables

The Dialogic® VoiceXML browser supports the following VoiceXML 2.0 session variables:

- ◆ session.telephone.ani
- ◆ session.telephone.dnis
- ◆ session.telephone.rdnis (always set to NULL)
- ◆ session.telephone.redirect\_reason

In addition to the above, other session variables are supported so that a VoiceXML script can access a complete set of information from the SIP call. Some of the new session variables are defined in VoiceXML 2.0, and others are Dialogic® extensions as noted on the following page.

### VoiceXML 2.0 Recommendations

session.connection.local.uri

Value: Set to the SIP Request-URI; note that any SIP escapes (%xx) in the Request-URI are still in this variable as well.

session.connection.remote.uri

Value: Set to the SIP From-URI.

session.connection.protocol.name

Value: Always SIP.

session.connection.protocol.version

Value: Always 2.0.

### Dialogic® Extensions

session.connection.media

An array where each element represents an RTP media stream. Each array element has the following associated attributes:

- ◆ type - indicates the MIME type of media stream (currently either "audio" or "video")
- ◆ subtype - indicates the MIME sub-type, or encoding, of the media stream (e.g. PCMU, PCMA, H264, etc.)

session.connection.protocol.sip.parameter

Value: An array of name/value pairs parsed from the SIP Request-URI; the name and value are "UnEscaped," meaning any SIP escapes (%xx) in the URI are expanded before being assigned.

session.connection.protocol.sip.parameter[n].name

Value: The name of the nth parameter.

session.connection.protocol.sip.parameter[n].value

Value: The "value" (if any, can be empty) of the nth parameter.

session.connection.protocol.sip.uri

Value: Set to the SIP Request-URI. Note that any SIP escapes (%xx) in the Request-URI are still in this variable as well. This variable duplicates the session.connection.local.uri by design.

session.connection.protocol.sip.to

Value: Set to the SIP To-URI. Since the VoiceXML 2.0 defined session.connection.local.uri variable is mapped to the request URI a separate variable is required to contain the contents of the SIP To header.

session.connection.protocol.sip.from

Value: Set to the SIP From-URI. This variable duplicates the session.connection.remote.uri by design.

session.connection.protocol.sip.call\_id

Value: Set to the SIP Call ID.

## Example

The following example shows the mapping between SIP headers and the newly supported VoiceXML session variables.

Given the following SIP INVITE request:

```
INVITE
sip:dialog@10.102.4.26;voicexml=http://10.102.4.9:9020
  /ivr/sip_init.vxml%3fdnis%3d961234567%26hasvideo%3d
  1;dogs=nice;user=phone
SIP/2.0
From: <sip:968037431@10.102.4.45:5060>;tag=2d04660a-
  13c4-40ed433f-9ec19fd-1786
To: <sip:961234567@10.102.4.9>
Call-ID: 6a7b774-2d04660a-13c4-40ed433f-9ec19fd-
  7061@10.102.4.45
CSeq: 1 INVITE
Via: SIP/2.0/UDP
  10.102.4.134:5060;branch=z9hG4bKi8MKi8i!yi8MK2SsMu8
  Uyake2q0OUqMi8i!y.1-1d9515c
Via: SIP/2.0/UDP
  10.102.4.45:5060;received=10.102.4.45;branch=z9hG4b
  K-40ed433f-9ec19fd-847
Max-Forwards: 69
User-Agent: RADVision ViaIP GW Vers. 2.0
Call-Info: <Media:Video>;purpose=info
Contact: <sip:968037431@10.102.4.45:5060>
Content-Length: 0
```

The IP Media Server populates the new VoiceXML session variables, as shown:

```
session.connection.protocol.name = "SIP"
session.connection.protocol.version = "2.0"
session.connection.local.uri =
    sip:dialog@10.102.4.26;voicexml=http://10.102.4.9:9
    020/ivr/sip_init.vxml%3fdnis%3d961234567%26hasvideo
    %3dl;dogs=nice;user=phone"
session.connection.remote.uri =
    <sip:968037431@10.102.4.45:5060>;tag=2d04660a-13c4-
    40ed433f-9ec19fd-1786
session.connection.protocol.sip.parameter[0].name =
    "voicexml"
session.connection.protocol.sip.parameter[0].value =
    "http://10.102.4.9:9020/ivr/sip_init.vxml?dnis=9612
    34567&hasvideo=1"
session.connection.protocol.sip.parameter[1].name =
    "dogs"
session.connection.protocol.sip.parameter[1].value =
    "nice"
session.connection.protocol.sip.parameter[2].name =
    "user"
session.connection.protocol.sip.parameter[2].value =
    "phone"
session.connection.protocol.sip.uri =
    sip:dialog@10.102.4.26;voicexml=http://10.102.4.9:9
    020/ivr/sip_init.vxml%3fdnis%3d961234567%26hasvideo
    %3dl;dogs=nice;user=phone"
session.connection.protocol.sip.to =
    <sip:961234567@10.102.4.9>
session.connection.protocol.sip.from =
    <sip:968037431@10.102.4.45:5060>;tag=2d04660a-13c4-
    40ed433f-9ec19fd-1786
session.connection.protocol.sip.call_id = 6a7b774-
    2d04660a-13c4-40ed433f-9ec19fd-7061@10.102.4.45
```

## Text Overlay Functionality

This section provides information on text overlay functionality:

- ◆ [Text Overlay Overview](#)
- ◆ [Characteristics of Text Overlay](#)
- ◆ [Text Overlay Restrictions](#)
- ◆ [Installing Font Files](#)
- ◆ [Dynamic Fonts](#)
- ◆ [Cached Fonts](#)
- ◆ [Text Overlay Demos](#)

### Text Overlay Overview

Dialogic® IP Media Server provides text overlay functionality that can be used to create interactive menus or to display corporate taglines, among other things. Text overlay functionality is supported for play only; it is not supported for record. Up to 16 simultaneous text overlays per RTP stream are supported.

Text overlay functionality is implemented in VoiceXML 2.0 using the `<overlay>`, `<content>`, `<font>`, and `<scroll>` elements. Default values are provided for various characteristics of the text overlay, such as font, font size, and font color. For reference information, see the `<overlay>`, `<content>`, `<font>`, and `<scroll>` element descriptions in this chapter. For examples of scripts, see “Text Overlay” (page 304) in Chapter 7, “Sample Code and Call Flows”.

The creation of a text overlay requires a definition of the overlay, such as its size and position, and a definition of the content to be displayed in the overlay.

The text overlay content definition provides the text string to be rendered and a font definition. The IP Media Server offers cached font and dynamic font rendering.

Cached font rendering uses the font definitions found in the `trcr.cfg` configuration file to provide fast rendering. These cached fonts require less processing at runtime, but require fonts and sizes to be identified at startup time. The character set supported by cached fonts is limited to ASCII characters. For more information, see “Cached Fonts” (page 238).

Dynamic font rendering allows rendering in any size using a font file that the application specifies at runtime. While it is more flexible, dynamic font rendering requires more processing at runtime. For more information, see “Dynamic Fonts” (page 237).

### Characteristics of Text Overlay

The following characteristics of a text overlay are configurable:

- ◆ Fonts, font sizes, and font colors for compatibility with varying video output formats. See `<font>`.

- ◆ The degree of transparency of text foreground (the text itself). See `<font fgcolor>`.
- ◆ The degree of transparency of text background (the space in between the letters). See `<overlay bgcolor>`.
- ◆ The size, position, and border of the text window in which the text input is displayed. See [<overlay>](#).
- ◆ Scrolling text, its direction, speed, and mode of display. See [<scroll>](#).
- ◆ Character-encoding, word wrap, and text alignment. See [<content>](#).

## Text Overlay Restrictions

The following restrictions are in effect for text overlay:

- ◆ For scrolling text, when scroll mode is continuous, there is no delay or space between the beginning of content and the end of content. To avoid this restriction, you can manually add spaces at the end of a text string.
- ◆ For scrolling text, the entire text overlay window scrolls. This means that the window is not fixed; instead, it scrolls with the text.
- ◆ For cached fonts, the font style must match the font file defined in `trcr.cfg`. For example, if the font file specifies a bold font, you must set the style to bold. If it specifies italics, you must set the style to italics.

## Installing Font Files

Fonts must be installed on the IP Media Server system that will execute the VoiceXML script. A distributed environment, such as one using Font Server, is not supported.

The IP Media Server requires that the Red Hat Linux Liberation font package be installed during the operating system install. Other font packages such as Hindi and Chinese are provided by Red Hat and may be installed as needed using the `rpm-Uvh <font package name>` command. Font packages are placed in `/usr/share/fonts/` directories. For more information on installation, see the Dialogic® *Software Quick Start Guide*.

To use fonts not provided by Red Hat, you must manually copy the font file such as `gz` or `rpm` to the directory of choice. After the font file is on your system, you can specify it using the `<font>` element. See [Dynamic Fonts](#) and [Cached Fonts](#) for more information.

## Dynamic Fonts

Dynamic font rendering allows rendering in any size using a font file that the application specifies at runtime. Dynamic font rendering is more flexible, but requires more processing at runtime and may add to play latency.

To use UTF-8 or GB18030 character encodings in a text overlay, you must use dynamic fonts. Cached fonts support the ASCII character set only.

To use dynamic fonts:

- 1 Ensure that the font files for the fonts to be used in the text overlay are already installed on your IP Media Server system. See [“Installing Font Files” \(page 237\)](#).
- 2 In the `<content>` element, specify a local file for `src` attribute. Specify values for other attributes as needed. For example:

```
<content src="file:///opt/snowshore/htdocs/vxml/demos/media/pizza.gb"
  encoding="GB18030" halign="center" >
```

- 3 In the `<font>` element, specify the font file and its location for `name` attribute. The font file must be local to the IP Media Server system. Specify values for other attributes as needed. For example:

```
<font name="font:///usr/share/fonts/chinese/TrueType/uming.ttf"
  size="30" fgcolor="black" />
```

## Cached Fonts

Cached font rendering uses the fonts defined in the `trcr.cfg` configuration file to provide fast rendering. These cached fonts require less processing at runtime, but require fonts and sizes to be identified and stored in memory when the IP Media Server is started. Play latency is typically reduced but more memory is consumed.

The character set supported by cached fonts is limited to ASCII characters. For other character encodings such as UTF-8 or GB18030, you must use dynamic fonts; see [“Dynamic Fonts” \(page 237\)](#).

To use cached fonts:

- 1 Ensure that the font files for the fonts to be used in the text overlay are already installed on your IP Media Server system. See [“Installing Font Files” \(page 237\)](#).
- 2 Define the fonts in the `[fonts]` section of `trcr.cfg`, located in `opt/snowshore/etc`. See [“trcr.cfg Description” \(page 238\)](#). After making changes in `trcr.cfg`, you must restart the IP Media Server.
- 3 In the `<content>` element, specify a local file for `src` attribute or inline ASCII text. Specify values for other attributes as needed.
- 4 In the `<font>` element, specify a font name defined in `trcr.cfg`. Specify values for other attributes as needed. For example:

```
<font name="SampleFont" size="24" fgcolor="blue" />
```

## trcr.cfg Description

The `trcr.cfg` configuration file contains font definitions used for cached font rendering. It contains two sections: `[global]` and `[fonts]`.

The [global] section of trcr.cfg specifies overall rendering behavior and contains the following parameters:

<b>[global] section parameters</b>	<b>Description</b>
debuglogmask	<p>Set of logging bits that can be activated to obtain detailed tracing information of render operations.</p> <p>Default: 0x00000003 (trace any detected WARNING or ERROR condition).</p> <p>This variable should not require any modification for standard operations.</p>
SharedMem	<p>Sets the size of the shared memory area that is used to pass rendered overlay content between the rendering process and the media server.</p> <p>This value is set at download time based on the number of transcoding instances in the license. A default value of 99 KB is allocated for each transcoding device. The value of 99 KB is determined by allotting for one overlay covering a QCIF resolution image:</p> <p>176 pixels wide * 144 pixels high * 4 bytes per pixel for rendering = 101376 bytes / 1024 = 99 kilobytes</p> <p>If necessary, you can change this value using the KBsPerOverlayResource variable in /opt/snowshore/etc/snowshore.cfg. The amount of memory is allocated using this formula: the sum of transcoding devices * KBsPerOverlayResource.</p> <p>You will need to increase the KBsPerOverlayResource value if you receive the following error in mserv.log:</p> <pre>trcr request failed (error=out of shared memory)</pre>

The [fonts] section of trcr.cfg specifies pre-defined fonts and contains the following parameters:

<b>[fonts] section parameters</b>	<b>Description</b>
Name	A short name to identify the font.
Font File Name	The name of the true type font file describing the font.
Pre-rendered From To	A range of font sizes to be pre-rendered.
Font sizes increment	The incremental value in the range of font sizes.

## Sample trcr.cfg

The following example shows font definitions in the [fonts] section of trcr.cfg.

```
[fonts]
#
#Name          Font File Name          Pre-rendered font sizes
#             #             From To             Increment
#-----
font MonoBold  "/usr/share/fonts/liberation/LiberationMono-Bold.ttf"    10  24    2
font MonoItal  "/usr/share/fonts/liberation/LiberationMono-Italic.ttf"   10  24    2
font Mono      "/usr/share/fonts/liberation/LiberationMono-Regular.ttf"  10  24    2
font SansBold  "/usr/share/fonts/liberation/LiberationSans-Bold.ttf"     10  24    2
font SansItal  "/usr/share/fonts/liberation/LiberationSans-Italic.ttf"   10  24    2
font Sans      "/usr/share/fonts/liberation/LiberationSans-Regular.ttf"  10  24    2
```

## Text Overlay Demos

Text overlay demos (sample applications) are provided in `/opt/snowshore/htdocs/vxml/demos/tovl`. For more information, see the demo readme file included in this directory.



## 7 - Sample Code and Call Flows

This chapter provides examples and call flow diagrams for a variety of scenarios.

This chapter includes the following sections:

- ◆ [Announcements](#)
- ◆ [Conferences](#)
- ◆ [IVR with MSCML](#)
- ◆ [VoiceXML](#)

# Announcements

## Play an Announcement as Early Media

The numbers in the left-hand column reference the Call Flow diagram in [Figure 8 \(page 244\)](#).

### Example 20. Announcement as Early Media

```
1          INVITE
           sip:annc@192.168.12.155:5060;early=yes
           ;play=
           file:///opt/
           snowshore/prompts/generic/10.ulaw
           SIP/2.0
Via: SIP/2.0/UDP 192.168.1.150:6100
To: <sip:annc@192.168.12.155:5060>
From: <sip:test0@192.168.12.153:5060>
Call-ID: 27125@192.168.1.150
Contact: sip:192.168.1.150:6100
CSeq: 2 INVITE
Content-Type: application/sdp
Supported: timer
Supported: 100rel
Session-Expires: 60
Content-Length: 153

v=0
o=SnowShoreUaV1 14250 3757 IN IP4
  192.168.1.150
s=SnowShore Sdp
t=0 0
m=audio 6000/1 RTP/AVP 0
c=IN IP4 192.168.12.154
a=sendrecv
a=ptime:20

2          SIP/2.0 100 Trying
Contact: sip:192.168.12.155:5060
Via: SIP/2.0/UDP 192.168.1.150:6100
To:
  <sip:annc@192.168.12.155:5060>;tag=100
  5580780
From: <sip:test0@192.168.12.153:5060>
Call-ID: 27125@192.168.1.150
CSeq: 2 INVITE
Content-Length: 0
```

## Example 20. Announcement as Early Media (continued)

```
3 SIP/2.0 183 Session Progress
Contact: sip:192.168.12.155:5060
Via: SIP/2.0/UDP 192.168.1.150:6100
To:
    <sip:annc@192.168.12.155:5060>;tag=100
    5580780
From: <sip:test0@192.168.12.153:5060>
Call-ID: 27125@192.168.1.150
CSeq: 2 INVITE
Content-Type: application/sdp
Require: 100rel
RSeq: 22752
Content-Length: 153

v=0
o=SnowShoreUaV1 27736 410 IN IP4
    192.168.12.155
s=SnowShore Sdp
t=0 0
m=audio 4362/1 RTP/AVP 0
c=IN IP4 192.168.12.156
a=sendrecv
a=ptime:20

4 SIP/2.0 200 OK
Via: Sip/2.0/UDP 192.168.1.150:6100
To:
    <sip:annc@192.168.12.155:5060>;tag=100
    5580780
From: <sip:test0@192.168.12.153:5060>
Call-ID: 27125@192.168.1.150
CSeq: 3 PRACK
Content-Length: 0

5 SIP/2.0 487 Request Terminated
Contact: sip:192.168.12.155:5060
Via: SIP/2.0/UDP 192.168.12.150:6100
To:
    sip:annc@192.168.12.155:5060;;tag=1005
    580780
From: sip:4444@192.168.12.153:5060
Call-ID: 27125@192.168.1.150
CSeq: 2 INVITE
Content-Length: 0
```

Example 20. Announcement as Early Media (continued)

```

6      ACK sip:annc@192.168.12.155:5060 SIP/2.0
      Via: SIP/2.0/UDP 192.168.12.150:6100
      To:
          sip:annc@192.168.12.155:5060>;tag=1005
          580780
      From: sip:4444@192.168.12.153:5060
      Call-ID: 27125@192.168.1.150
      CSeq: 2 ACK
      Content-Length: 0
    
```

## Call Flow for an Early Media Announcement

The following illustrates a call flow diagram for early media announcement.

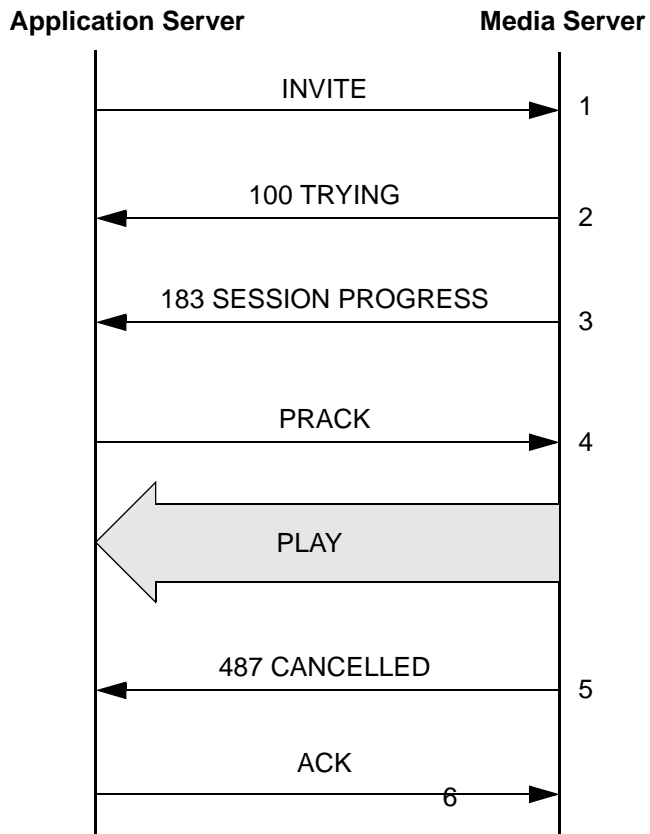


Figure 8. Call Flow: Announcement, Early Media

## Playing an Announcement as Normal Media

The numbers in the left-hand column reference the diagram [Figure 9](#) (page 247).

### Example 21. Playing an Announcement as Normal Media

```

1      sip:annc@192.168.12.155:5060;play=file:///
      /opt/snowshore/
      prompts/generic/10.ulaw;early=no
      SIP/2.0
Via: SIP/2.0/UDP 192.168.1.150:6100
To: <sip:annc@192.168.12.155:5060>
From: <sip:test0@192.168.12.153:5060>
Call-ID: 27103@192.168.1.150
Contact: sip:192.168.1.150:6100
CSeq: 2 INVITE
Content-Type: application/sdp
Supported: timer
Supported: 100rel
Session-Expires: 60
Content-Length: 154

      v=0
      o=SnowShoreUaV1 11845 20808 IN IP4
      192.168.1.150
      s=SnowShore Sdp
      t=0 0
      m=audio 6000/1 RTP/AVP 0
      c=IN IP4 192.168.12.154
      a=sendrecv
      a=ptime:20

2      SIP/2.0 100 Trying
      Contact: sip:192.168.12.155:5060
      Via: SIP/2.0/UDP 192.168.1.150:6100
      To:
      <sip:annc@192.168.12.155:5060>;tag=1005
      580654
      From: <sip:test0@192.168.12.153:5060>
      Call-ID: 27103@192.168.1.150
      CSeq: 2 INVITE
      Content-Length: 0

```

## Example 21. Playing an Announcement as Normal Media (continued)

```
3 SIP/2.0 200 OK
  Contact: sip:192.168.12.155:5060
  Via: SIP/2.0/UDP 192.168.1.150:6100
  To:
    <sip:annc@192.168.12.155:5060>;tag=1005
    580654
  From: <sip:test0@192.168.12.153:5060>
  Call-ID: 27103@192.168.1.150
  CSeq: 2 INVITE
  Content-Type: application/sdp
  Session-Expires: 60
  Require: timer
  Content-Length: 152

  v=0
  o=SnowShoreUaV1 5987 313 IN IP4
    192.168.12.155
  s=SnowShore Sdp
  t=0 0
  m=audio 4360/1 RTP/AVP 0
  c=IN IP4 192.168.12.156
  a=sendrecv
  a=ptime:20

4 ACK sip:annc@192.168.12.155:5060 SIP/2.0
  Via: SIP/2.0/UDP 192.168.1.150:6100
  To:
    <sip:annc@192.168.12.155:5060>;tag=1005
    580654
  From: <sip:test0@192.168.12.153:5060>
  Call-ID: 27103@192.168.1.150
  CSeq: 2 ACK
  Content-Length: 0

5 BYE sip:annc@192.168.12.150:5060 SIP/2.0
  Via: SIP/2.0/UDP 192.168.1.150:6100
  To:
    <sip:test0@192.168.12.153:5060>;tag=100
    5580654
  From: <sip:annc@192.168.12.155:5060>
  Call-ID: 27103@192.168.1.150
  CSeq: 3 BYE
  Supported: timer
  Supported: 100rel
  Content-Length: 0
```

## Example 21. Playing an Announcement as Normal Media (continued)

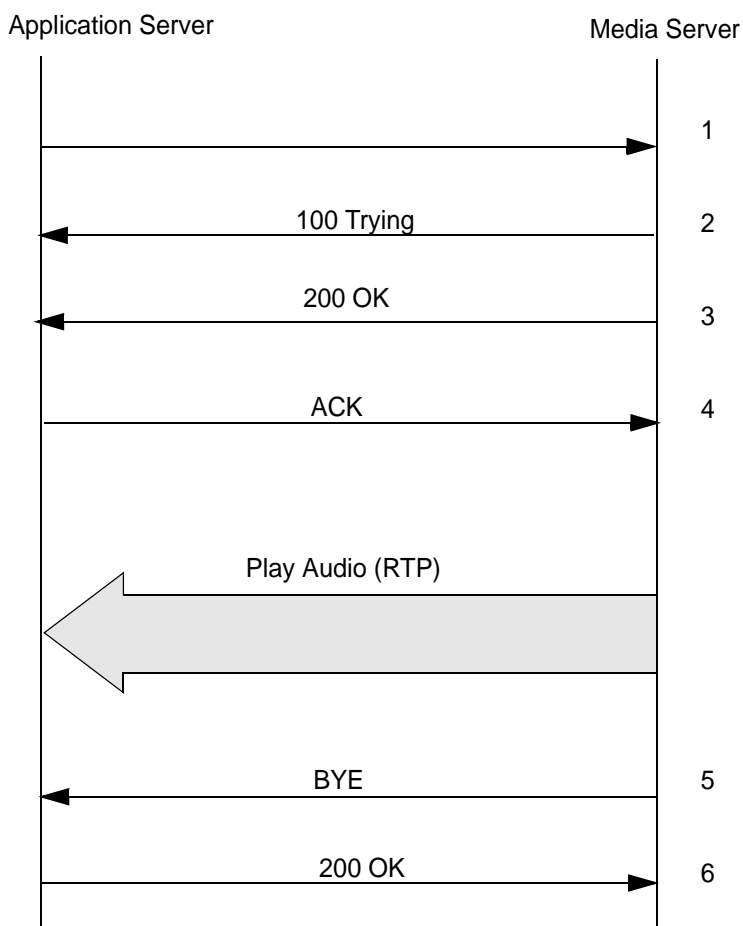
```

6      SIP/2.0 200 OK
      Via: SIP/2.0/UDP 192.168.1.150:6100
      To:
        <sip:test0@192.168.12.153:5060>;tag=100
        5580654
      From: <sip:annc@192.168.12.155:5060>
      Call-ID: 27103@192.168.1.150
      CSeq: 3 BYE
      Content-Length: 0

```

## Call Flow for a Normal Media Announcement

The following illustrates a call flow diagram for normal media announcement.



**Figure 9. Call Flow: Announcement, Normal Media**

## Stopping Media—Hold

A SIP INVITE request places the remote party on hold.

### Example 22. Stopping Media Using Hold

---

Note: Hold is represented with 0.0.0.0 on the line beginning with c=

---

[SIP header is not shown.]

```
v=0
o=anonymous 0 0 IN IP4 10.128.41.9
s=anonymous0
t=0 0
m=audio 5010 RTP/AVP 0 101
c=IN IP4 0.0.0.0
a=rtpmap:0 pcmu/8000
a=sendrecv
a=ptime:20
a=rtpmap:101 telephone-event/8000/1
m=video 5012 RTP/AVP 34
c=IN IP4 0.0.0.0
a=rtpmap:34 h263/90000
a=sendrecv
```

A successful response (200 OK) to the hold INVITE request confirms the on-hold state. There is nothing unique in the SDP.

```
200 OK
v=0
o=SnowShoreUaV1 846930886 1681692777 IN IP4
  10.128.41.223
s=SnowShore Sdp

t=0 0
m=audio 6000 RTP/AVP 0 101
c=IN IP4 10.128.41.223
a=sendrecv
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000/1
m=video 6002 RTP/AVP 34
c=IN IP4 10.128.41.223
a=sendrecv
a=ptime:20
a=rtpmap:34 H263/90000
```

re-INVITE to release the hold

[SIP header not shown]



---

```
v=0
o=anonymous 0 0 IN IP4 10.128.41.9
s=anonymous0
t=0 0
m=audio 5010 RTP/AVP 0 101
c=IN IP4 10.128.41.9
a=rtpmap:0 pcmu/8000
a=sendrecv
a=ptime:20
a=rtpmap:101 telephone-event/8000/1
m=video 5012 RTP/AVP 34
c=IN IP4 10.128.41.9
a=rtpmap:34 h263/90000
a=sendrecv
```

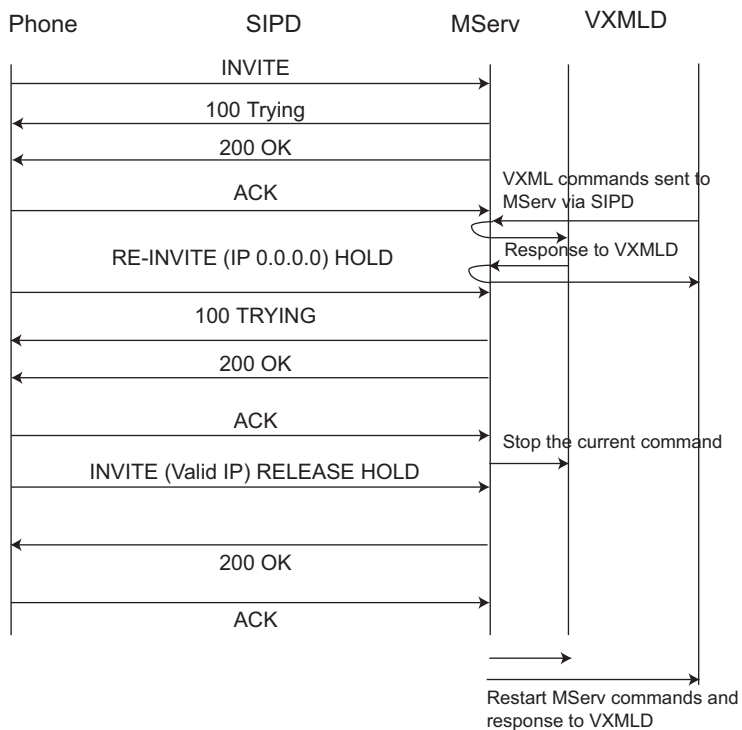
200 OK sent by sipd to release the HOLD.

[SIP header not shown]

```
v=0
o=SnowShoreUaV1 846930886 1681692777 IN IP4
 10.128.41.223
s=SnowShore Sdp
t=0 0
m=audio 0 RTP/AVP 0 101
c=IN IP4
a=sendrecv
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000/1
m=video 5012 RTP/AVP 34
c=IN IP4 10.128.41.9
a=sendrecv
a=ptime:20
a=rtpmap:34 H263/90000
```

## Call Flow for Stopping Media—Hold

The following illustrates a call flow diagram for stopping media using hold.



**Figure 10. Call Flow: Stopping Media Using Hold**

## Repeating the Audio

The following illustrates sample code for repeating audio indefinitely.

### Example 23. Repeating Audio Indefinitely

```
<request>
<play id=21212>
<prompt baseurl="file:///opt/snowshore/prompts/
generic/" locale="en_US" repeat="infinite"
duration="infinite" offset="0" delay="2000">
  <audio url="num_changed.wav" />
  <audio url="new_number.wav" />
  <variable type="dig" subtype="ndn"
value="9725551212" />
  <audio url="make_note.wav" />
</prompt>
</play>
</request>
```

## Conferences

### Creating a Simple Conference

#### Example 24. Simple Conference

```
INVITE sip:conf=1234@192.168.12.153 SIP/2.0
From: sip:threepcc@192.168.1.126;tag=1as8ut
To: sip:conf=1234@192.168.12.153
Call-ID: 1031579120515@192.168.1.126
CSeq: 1327320033 INVITE
Content-Length: 201
Content-Type: application/sdp
Contact: <sip:192.168.1.126:5060;
transport=udp>
Via: SIP/2.0/UDP 192.168.1.126:5060
```

```
v=0
o=Pingtel 5 5 IN IP4 192.168.12.109
s=phone-call
c=IN IP4 192.168.12.109
t=0 0
m=audio 8770 RTP/AVP 0 96 8
a=rtpmap:0 pcmu/8000/1
a=rtpmap:96 telephone-event/8000/1
a=rtpmap:8 pcma/8000/1
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
```

```
To: sip:conf=1234@192.168.12.153;tag=
1031665657
From: sip:threepcc@192.168.1.126;tag=1as8ut
Call-ID: 1031579120515@192.168.1.126
CSeq: 1327320033 INVITE
Contact: sip:192.168.12.153:5060
Content-Type: application/sdp
Session-Expires: 120
Content-Length: 153
a=ptime:20
```

```
v=0
o=SnowShoreUaV1 22263 30720 IN IP4 192.168.12.153
s=SnowShore Sdp
t=0 0
m=audio 4202 RTP/AVP 0
c=IN IP4 192.168.12.154
a=sendrecv
```

## Call Flow for a Simple Conference (Normal Media)

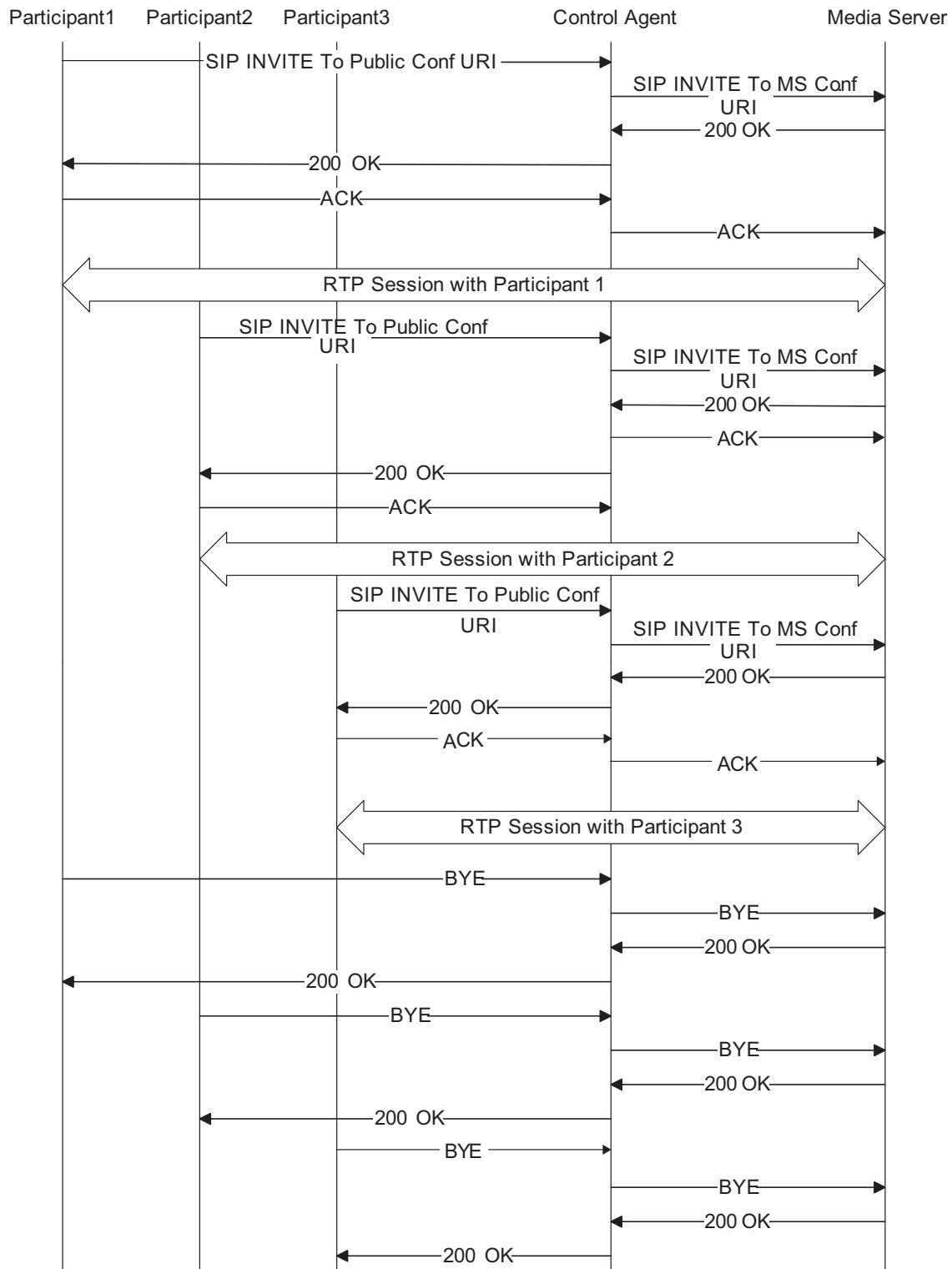


Figure 11. Call Flow: Simple Conference

---

## Creating an Advanced Conference

### Example 25. Advanced Conference

```
INVITE sip:conf=1234@192.168.12.153 SIP/2.0
From: sip:mscmltest@192.168.1.126;tag=
      7hl229
To: sip:conf=1234@192.168.12.153
Call-ID: 1031579500288@192.168.1.126
CSeq: 1597127738 INVITE
Content-Length: 504
Content-Type: multipart/mixed;boundary=snow-bound
Contact: <sip:192.168.1.126:5060;transport=
      udp>
Via: SIP/2.0/UDP 192.168.1.126:5060

--snow-bound
Content-type:application/mediaservercontrol+xml

<?xml version='1.0'?>
<MediaServerControl version='1.0'>
<request>
<configure_conference reserveconfmedia='yes'
      reservedtalkers='10' />
</request>
</MediaServerControl>
--snow-bound
Content-type:application/sdp

v=0
o=Pingtel 5 5 IN IP4 192.168.12.109
s=phone-call
c=IN IP4 0.0.0.0
t=0 0

--snow-bound--
```

---

**Note:** In compliance with RFC1341 (MIME), there is a blank line following the boundary marker and a CRLF separating each line in the multipart/mixed MSCML body.

---

### Expected Response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
To: sip:conf=1234@192.168.12.153;tag=
      1031666037
From: sip:mscmltest@192.168.1.126;tag=
      7hl229
Call-ID: 1031579500288@192.168.1.126
CSeq: 1597127738 INVITE
```

---

```
Contact: sip:192.168.12.153:5060
Content-Type: multipart/mixed; boundary=
    snow-boundary-1
Session-Expires: 120
Content-Length: 444

--snow-boundary-1
Content-Type: application/mediaservercontrol+xml

<?xml version="1.0"?>
<MediaServerControl version="1.0">
    <response request="configure_conference"
        code="200" text="OK"/>
</MediaServerControl>

--snow-boundary-1
Content-Type: application/sdp

v=0
o=SnowShoreUaV1 15485 15497 IN IP4 192.168.12.153
s=SnowShore Sdp
t=0 0
c=IN IP4 0.0.0.0

--snow-boundary-1--
```

## Call Flow to Set up an Advanced Conference

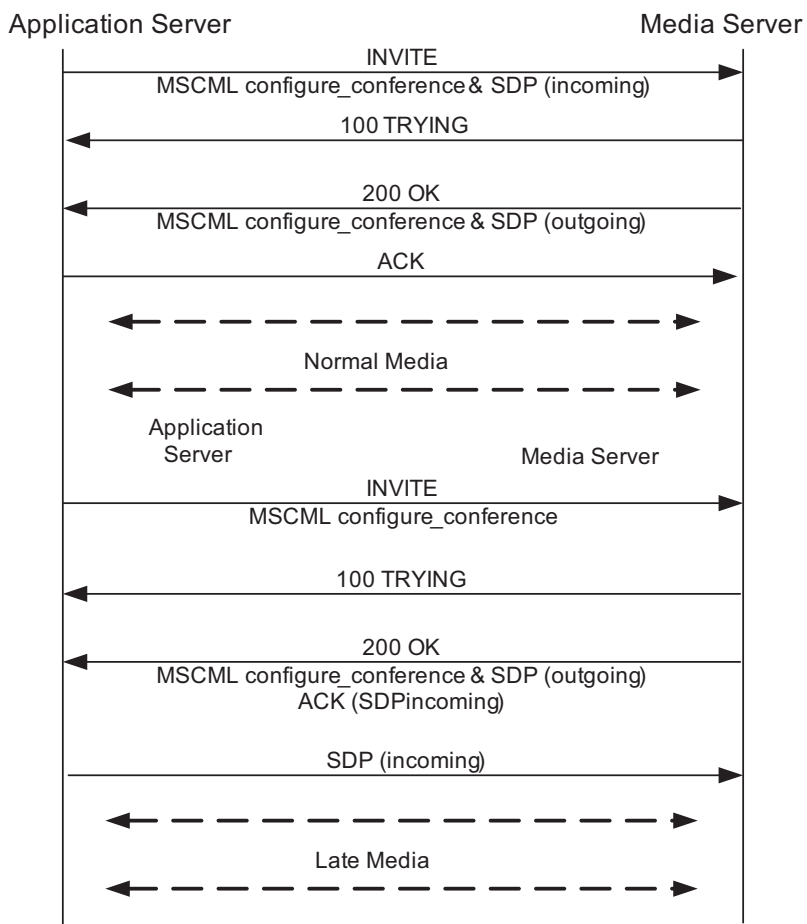


Figure 12. Call Flow: Advanced Conference



---

## Modifying Conference Using Subscribe

### Example 26. Modifying a Conference Using Subscribe

```
INFO sip:conf=1234@192.168.12.153 SIP/2.0
From: sip:mscmltest@192.168.1.126;tag=
      7hl229
To: sip:conf=1234@192.168.12.153
Call-ID: 1031579500288@192.168.1.126
CSeq: 1597127738 INFO
Content-Length: 504
Content-Type: application/mediaservercontrol+xml
Contact: <sip:192.168.1.126:5060;transport=
      udp>
Via: SIP/2.0/UDP 192.168.1.126:5060

<?xml version='1.0'?>
<MediaServerControl version="1.0">
  <request>
    <configure_conference>
      <subscribe>
        <events>
          <activetalkers report="yes"
        </events>
      </subscribe>
    </configure_conference>
  </request>
</MediaServerControl>
```

---

## Providing Communication for Participant in an Advanced Conference

---

Note: This example uses defaults.

---

### Example 27. Adding Participant to Advanced Conference

```
INVITE sip:conf=1234@192.168.12.153 SIP/2.0
From: sip:threepcc@192.168.1.126;tag=jtopyd
To: sip:conf=1234@192.168.12.153
Call-ID: 1031580002825@192.168.1.126
CSeq: 2076031605 INVITE
Content-Length: 201
Content-Type: application/sdp
Contact: <sip:192.168.1.126:5060;transport=
  udp>
Via: SIP/2.0/UDP 192.168.1.126:5060

v=0
o=Pingtel 5 5 IN IP4 192.168.12.109
s=phone-call
c=IN IP4 192.168.12.109
t=0 0
m=audio 8770 RTP/AVP 0 96 8
a=rtpmap:0 pcmu/8000/1
a=rtpmap:96 telephone-event/8000/1
a=rtpmap:8 pcma/8000/1
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
To: sip:conf=1234@192.168.12.153;tag=
  1
  031666539
From: sip:threepcc@192.168.1.126;tag=
  jtopyd
Call-ID: 1031580002825@192.168.1.126
CSeq: 2076031605 INVITE
Contact: sip:192.168.12.153:5060
Content-Type: application/sdp
Session-Expires:120
Content-Length: 152

v=0
o=SnowShoreUaV1 7030 13061 IN IP4 192.168.12.153
s=SnowShore Sdp
t=0 0
m=audio 4208 RTP/AVP 0
c=IN IP4 192.168.12.154
a=sendrecv
a=ptime:20
```

---

## Joining a Participant Using Special Attributes

### Example 28. Joining Participant Using Special Attributes

```
INVITE sip:conf=1234@9.9.3.1 SIP/2.0
From: sip:confbridge@10.10.10.150;tag=4qjtjc
To: sip:conf=1234@9.9.3.1
Call-ID: 1031256911282@10.10.10.150
CSeq: 1887335379 INVITE
Content-Length: 481
Content-Type: multipart/mixed;
    boundary=confbridge-bound
Contact: <sip:10.10.10.150:5060;
    transport=udp>
Via: SIP/2.0/UDP 10.10.10.150:5060

--confbridge-bound
content-type:application/sdp

v=0
o=Pingtel 5 5 IN IP4 10.10.10.16
s=phone-call
c=IN IP4 10.10.10.16
t=0 0
m=audio 8766 RTP/AVP 0 96 8
a=rtpmap:0 pcmu/8000/1
a=rtpmap:96 telephone-event/8000/1
a=rtpmap:8 pcma/8000/1
--confbridge-bound
content-type:application/mediaservercontrol+xml

<?xml version='1.0'?>
<MediaServerControl version='1.0'>
<request>
<configure_leg mixmode='private' />
</request>
</MediaServerControl>
--confbridge-bound--
```

## Suspending Communications within a Conference

### Example 29. Suspending Communications during a Conference (Continued)

```
INFO sip:192.168.12.153:5060 SIP/2.0
From: sip:confbridge@192.168.1.126;tag=oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407038 INFO
Content-Length: 142

Content-Type: application/mediaservercontrol+xml
Via: SIP/2.0/UDP 192.168.1.126:5060

<?xml version='1.0'?>
<MediaServerControl version='1.0'>
<request>
<configure_leg mixmode='mute' />
</request>
</MediaServerControl>

SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
To: sip:conf=1234@192.168.12.153;tag=
    1031666918
From: sip:confbridge@192.168.1.126;tag=
    oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407038 INFO
Content-Length: 0
```

---

## Response to Mute a Conference Participant

### Example 30. Response to Muting a Participant

```
INFO sip:confbridge@192.168.1.126:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.153:5060

To: sip:confbridge@192.168.1.126;tag=oi32zw
From: sip:conf=1234@192.168.12.153;tag=
      1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 15398 INFO
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 137
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_leg" code="200"
    text="OK" />
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.12.153:5060
From: sip:conf=1234@192.168.12.153;tag=
      1031666918
To: sip:confbridge@192.168.1.126;tag=oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 15398 INFO
Content-Length: 0
```

## Playing Audio to Conference Participant

### Changing Mixmode to Parked

Example 31. Changing Participant Mixmode to Parked

```
INFO sip:192.168.12.153:5060 SIP/2.0
From: sip:confbridge@192.168.1.126;tag=oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407032 INFO
Content-Length: 144
Content-Type: application/mediaservercontrol+xml
Via: SIP/2.0/UDP 192.168.1.126:5060
<?xml version='1.0'?>
<MediaServerControl version='1.0'>
<request>
<configure_leg mixmode='parked' />
</request>
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
To: sip:conf=1234@192.168.12.153;tag=
    1031666918

From: sip:confbridge@192.168.1.126;tag=
    oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407032 INFO
Content-Length: 0
```

## Response to Parked

```

INFO sip:confbridge@192.168.1.126:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.153:5060
To: sip:confbridge@192.168.1.126;tag=oi32zw
From: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 15392 INFO
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 137
<?xml version="1.0"?>
<MediaServerControl version="1.0">
    <response request="configure_leg" code="200"
        text="OK" />
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.12.153:5060
From: sip:conf=1234@192.168.12.153;tag=1031666918
To: sip:confbridge@192.168.1.126;tag=oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 15392 INFO
Content-Length: 0

```

## Playing the Audio

### Example 32. Playing Audio

```

INFO sip:192.168.12.153:5060 SIP/2.0
From: sip:confbridge@192.168.1.126;tag=
    oi32zw
To: sip:conf=1234@192.168.12.153;tag=
    1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407033 INFO
Content-Length: 194
Content-Type: application/mediaservercontrol+xml
Via: SIP/2.0/UDP 192.168.1.126:5060
<?xml version='1.0'?>
<MediaServerControl version='1.0'>
<request>
<play prompt='http://192.168.1.126:8013/
    confbridge/audio/help.raw' promptencoding='ulaw' />
</request>
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
To: sip:conf=1234@192.168.12.153;tag=
    1031666918

```

```
From: sip:confbridge@192.168.1.126;tag=
  oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407033 INFO
Content-Length: 0
```

### Response to Message to Play Audio

#### Example 33. Response to Message to Play Audio

```
INFO sip:confbridge@192.168.1.126:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.153:5060
To: sip:confbridge@192.168.1.126;tag=oi32zw
From: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 15393 INFO
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 160
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="play" code="200" text="OK"
    reason="EOF" playduration="8890" />
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.12.153:5060
From: sip:conf=1234@192.168.12.153;tag=1031666918
To: sip:confbridge@192.168.1.126;tag=oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 15393 INFO
Content-Length: 0
```

### Changing Participant Mixmode Back to Full

#### Example 34. Changing Mixmode Back to Full

```
INFO sip:192.168.12.153:5060 SIP/2.0
From: sip:confbridge@192.168.1.126;tag=
  oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407034 INFO
Content-Length: 142
Content-Type: application/mediaservercontrol+xml
Via: SIP/2.0/UDP 192.168.1.126:5060
<?xml version='1.0'?>
<MediaServerControl version='1.0'>
<request>
```



```

<configure_leg mixmode='full' />
</request>
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060

To: sip:conf=1234@192.168.12.153;tag=
    1031666918
From: sip:confbridge@192.168.1.126;tag=
    oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407034 INFO
Content-Length: 0

```

## Response to Mixmode Change

### Example 35. Response to Mixmode Change

```

INFO sip:confbridge@192.168.1.126:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.153:5060
To: sip:confbridge@192.168.1.126;tag=oi32zw
From: sip:conf=1234@192.168.12.153;tag=
    1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 15394 INFO
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 137
<?xml version="1.0"?>
<MediaServerControl version="1.0">
<response request="configure_leg" code="200" text="OK"
    />
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.12.153:5060
From: sip:conf=1234@192.168.12.153;tag=
    1031666918
To: sip:confbridge@192.168.1.126;tag=oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 15394 INFO
Content-Length: 0

```

## Playcollect and Playrecord in a Conference

### Changing Participant Mixmode to Parked

Example 36. Changing Mixmode to Parked Mode

```
INFO sip:192.168.12.153:5060 SIP/2.0
From: sip:confbridge@192.168.1.126;tag=oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407032 INFO
Content-Length: 144
Content-Type: application/mediaservercontrol+xml
Via: SIP/2.0/UDP 192.168.1.126:5060
<?xml version='1.0'?>
<MediaServerControl version='1.0'>
<request>
<configure_leg mixmode='parked' />
</request>
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
To: sip:conf=1234@192.168.12.153;tag=
1031666918
From: sip:confbridge@192.168.1.126;tag=
oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407032 INFO
Content-Length: 0
```

### Response to Parked

Example 37. Response to Parked

```
INFO sip:confbridge@192.168.1.126:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.153:5060
To: sip:confbridge@192.168.1.126;tag=oi32zw
From: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 15392 INFO
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 137
<?xml version="1.0"?>
<MediaServerControl version="1.0">
<response request="configure_leg" code="200"
text="OK" />
</MediaServerControl>
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.12.153:5060
From: sip:conf=1234@192.168.12.153;tag=1031666918
To: sip:confbridge@192.168.1.126;tag=oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 15392 INFO
Content-Length: 0
```

## PlayCollect

```
INFO sip:192.168.12.153:5060 SIP/2.0
From: sip:confbridge@192.168.1.126;tag=oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407027 INFO
Content-Length: 237
Content-Type: application/mediaservercontrol+xml
Via: SIP/2.0/UDP 192.168.1.126:5060
<?xml version='1.0'?>
<MediaServerControl version='1.0'>
<request>
<playcollect
  prompt='http://192.168.1.126:8013/confbridge/audio/
askauthcode.raw' barge='yes' cleardigits='yes'
  maxdigits='4' />
</request>
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
From: sip:confbridge@192.168.1.126;tag=oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407027 INFO
Content-Length: 0
```

## Response to PlayCollect

```
INFO sip:confbridge@192.168.1.126:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.153:5060
To: sip:confbridge@192.168.1.126;tag=oi32zw
From: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 15387 INFO
Content-Type: application/mediaservercontrol+xml
Supported:timer
Supported:100rel
Content-Length: 183

<?xml version="1.0"?>
<MediaServerControl version="1.0">
```

```
        <response request="playcollect" code="200"
          text="OK" reason="timeout" digits="85"
          playduration="2260"/>
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.12.153:5060
From: sip:conf=1234@192.168.12.153;tag=1031666918
To: sip:confbridge@192.168.1.126;tag=oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 15387 INFO
Content-Length: 0
```

## Sending PlayRecord

```
INFO sip:192.168.12.153:5060 SIP/2.0
From: sip:confbridge@192.168.1.126;tag=oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407028 INFO
Content-Length: 300
Content-Type: application/mediaservercontrol+xml
Via: SIP/2.0/UDP 192.168.1.126:5060
<?xml version='1.0'?>
<MediaServerControl version='1.0'>
<request>
<playrecord
  prompt='http://192.168.1.126:8013/confbridge/audio/
  tellyourname.raw' promptencoding='ulaw' barge='yes'
  cleardigits='yes'
    recurl='file:///var/snowshore/rec/temp.ulaw'
  recencoding='ulaw' />
</request>
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
From: sip:confbridge@192.168.1.126;tag=oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407028 INFO
Content-Length: 0
```

## Response to PlayRecord

```
INFO sip:confbridge@192.168.1.126:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.153:5060
To: sip:confbridge@192.168.1.126;tag=oi32zw
From: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 15388 INFO
Content-Type: application/mediaservercontrol+xml
Supported:timer
Supported:100rel
```

```

Content-Length: 197
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="playrecord" code="200" text="OK"
    reason="digit" digits="#" reqlength="14320"
    playduration="4820" />
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.12.153:5060
From: sip:conf=1234@192.168.12.153;tag=1031666918
To: sip:confbridge@192.168.1.126;tag=oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 15388 INFO
Content-Length: 0

```

## Changing Participant Back to Full

```

INFO sip:192.168.12.153:5060 SIP/2.0
From: sip:confbridge@192.168.1.126;tag=oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407030 INFO
Content-Length: 142
Content-Type: application/mediaservercontrol+xml
Via: SIP/2.0/UDP 192.168.1.126:5060
<?xml version='1.0'?>
<MediaServerControl version='1.0'>
  <request>
  <configure_leg mixmode='full' />
  </request>
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.126:5060
From: sip:confbridge@192.168.1.126;tag=oi32zw
To: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 1066407030 INFO
Content-Length: 0

```

## Response to Full

```

INFO sip:confbridge@192.168.1.126:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.153:5060
To: sip:confbridge@192.168.1.126;tag=oi32zw
From: sip:conf=1234@192.168.12.153;tag=1031666918
Call-ID: 1031580381757@192.168.1.126
CSeq: 15390 INFO
Content-Type: application/mediaservercontrol+xml
Supported:timer
Supported:100rel
Content-Length: 137
<?xml version="1.0"?>

```

```
<MediaServerControl version="1.0">
  <response request="configure_leg" code="200"
    text="OK" />
</MediaServerControl>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.12.153:5060
From: sip:conf=1234@192.168.12.153;tag=1031666918
To: sip:confbridge@192.168.1.126;tag=oi32zw
Call-ID: 1031580381757@192.168.1.126
CSeq: 15390 INFO
Content-Length: 0
```

## Coached Conferencing

### Example 38. Coached Conferencing

#### 1 Invitation for control leg of enhanced conference

```
INVITE sip:conf=1234@192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5558;tag=1
To: sip:conf=1234@192.168.12.150:5060
Subject: Call from Control
Cseq: 1 INVITE
Call-ID: 1119358373
Contact: <sip:control@192.168.12.158:5558>
Content-Type: multipart/mixed;boundary=salem
Via: SIP/2.0/UDP 192.168.12.158:5558
```

```
--salem
Content-Type: application/sdp
```

```
v=0
o=mserv 1 1119358373 IN IP4 192.168.12.151
s=Riptide_Media_Server
c=IN IP4 0.0.0.0
t=0 0
m=audio 0 RTP/AVP 0
```

```
--salem
Content-Type: application/mediaservercontrol+xml
```

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_conference reservedtalkers="20"
      reserveconfmedia="no">
    </configure_conference>
  </request>
</MediaServerControl>
```

```
--salem--
```

## First response from IP Media Server:

```

SIP/2.0 100 Trying
Contact: sip:192.168.12.150:5060
Server: SnowShoreMediaServer/1.4.0-050614A
Via: SIP/2.0/UDP 192.168.12.158:5558
To: sip:conf=1234@192.168.12.150:5060;tag=1119358026
From: sip:control@192.168.12.158:5558;tag=1
Call-ID: 1119358373
CSeq: 1 INVITE
Content-Length: 0

```

## Second response from IP Media Server:

```

SIP/2.0 200 Ok
Server: SnowShoreMediaServer/1.4.0-050614A
Via: SIP/2.0/UDP 192.168.12.158:5558
To: sip:conf=1234@192.168.12.150:5060;tag=1119358026
From: sip:control@192.168.12.158:5558;tag=1
Call-ID: 1119358373
CSeq: 1 INVITE
Contact: sip:192.168.12.150:5060
Content-Type: multipart/mixed;boundary=snow-boundary-1
Session-Expires: 6000
Content-Length: 466

--snow-boundary-1
Content-Type: application/mediaservercontrol+xml

<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_conference"
    code="200" text="OK"/>
</MediaServerControl>
--snow-boundary-1
Content-Type: application/sdp

v=0
o=SnowShoreUaV1 1102520059 2044897763 IN IP4
  192.168.12.150
s=SnowShore Sdp
t=0 0
m=audio 0 RTP/AVP 0
c=IN IP4 0.0.0.0
a=sendrecv
a=ptime:20
a=rtpmap:0 PCMU/8000

--snow-boundary-1--

```

-----  
Customer Leg Invitation

```
INVITE sip:conf=1234@192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5560;tag=1
To: sip:conf=1234@192.168.12.150:5060
Subject: Call from Control
Cseq: 1 INVITE
Call-ID: 1119358621
Contact: <sip:control@192.168.12.158:5560>
Content-Type: multipart/mixed;boundary=salem
Via: SIP/2.0/UDP 192.168.12.158:5560
```

```
--salem
Content-Type: application/sdp
v=0
o=mserv 1 1119358621 IN IP4 192.168.12.151
s=Riptide_Media_Server
c=IN IP4 192.168.12.155
t=0 0
m=audio 8766 RTP/AVP 0
```

```
--salem
Content-Type: application/mediaservercontrol+xml
```

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg id="customer"/>
  </request>
</MediaServerControl>
```

```
--salem--
```

IP Media Server response to client application:

```
SIP/2.0 200 Ok
Server: SnowShoreMediaServer/1.4.0-050614A
Via: SIP/2.0/UDP 192.168.12.158:5560
To: sip:conf=1234@192.168.12.150:5060;
  tag=1119358274
From: sip:control@192.168.12.158:5560;tag=1
Call-ID: 1119358621
CSeq: 1 INVITE
Contact: sip:192.168.12.150:5060
Content-Type: multipart/mixed;boundary=
  snow-boundary-1
Session-Expires: 6000
Content-Length: 469

--snow-boundary-1
Content-Type: application/mediaservercontrol+xml
```



```

<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_leg" code="200"
    text="OK"/>
</MediaServerControl>
--snow-boundary-1
Content-Type: application/sdp

v=0
o=SnowShoreUaV1 1365180540 1540383426 IN IP4
  192.168.12.150
s=SnowShore Sdp
t=0 0
m=audio 6006 RTP/AVP 0
c=IN IP4 192.168.12.150
a=sendrecv
a=ptime:20
a=rtpmap:0 PCMU/8000
--snow-boundary-1--
-----

```

## 2 Agent Invite

```

INVITE sip:conf=1234@192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5562;tag=1
To: sip:conf=1234@192.168.12.150:5060
Subject: Call from Control
Cseq: 1 INVITE
Call-ID: 1119358826
Contact: <sip:control@192.168.12.158:5562>
Content-Type: multipart/mixed;boundary=salem
Via: SIP/2.0/UDP 192.168.12.158:5562

--salem
Content-Type: application/sdp

v=0
o=mserv 1 1119358826 IN IP4 192.168.12.151
s=Riptide_Media_Server
c=IN IP4 192.168.12.156
t=0 0
m=audio 8766 RTP/AVP 0

--salem
Content-Type: application/mediaservercontrol+xml
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg id="agent"/>
  </request>
</MediaServerControl>

```

--salem--

Response to agent:

```
SIP/2.0 200 Ok
Server: SnowShoreMediaServer/1.4.0-050614A
Via: SIP/2.0/UDP 192.168.12.158:5562
To: sip:conf=1234@192.168.12.150:5060;
    tag=1119358479
From: sip:control@192.168.12.158:5562;tag=1
Call-ID: 1119358826
CSeq: 1 INVITE
Contact: sip:192.168.12.150:5060
Content-Type: multipart/mixed;boundary=snow-boundary-1
Session-Expires: 6000
Content-Length: 467
--snow-boundary-1
Content-Type: application/mediaservercontrol+xml
```

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
    <response request="configure_leg" code="200"
        text="OK"/>
</MediaServerControl>
--snow-boundary-1
Content-Type: application/sdp
```

```
v=0
o=SnowShoreUaV1 1303455736 35005211 IN IP4
    192.168.12.150
s=SnowShore Sdp
t=0 0
m=audio 6008 RTP/AVP 0
c=IN IP4 192.168.12.150
a=sendrecv
a=ptime:20
a=rtpmap:0 PCMU/8000
```

--snow-boundary-1--

### 3 Supervisor- invited with ID and put into private mode

```
INVITE sip:conf=1234@192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5564;tag=1
To: sip:conf=1234@192.168.12.150:5060
Subject: Call from Control
Cseq: 1 INVITE
Call-ID: 1119359020
Contact: <sip:control@192.168.12.158:5564>
Content-Type: multipart/mixed;boundary=salem
Via: SIP/2.0/UDP 192.168.12.158:5564

--salem
```

```

Content-Type: application/sdp

v=0
o=mserv 1 1119359020 IN IP4 192.168.12.151
s=Riptide_Media_Server
c=IN IP4 192.168.12.157
t=0 0
m=audio 32126 RTP/AVP 0

--salem
Content-Type: application/mediaservercontrol+xml

<?xml version="1.0"?>
<MediaServerControl version="1.0">
<request>
    <configure_leg id="super" mixmode="private"/>
</request>
</MediaServerControl>

```

#### Response from Media Server:

```

SIP/2.0 200 Ok
Server: SnowShoreMediaServer/1.4.0-050614A
Via: SIP/2.0/UDP 192.168.12.158:5564
To: sip:conf=1234@192.168.12.150:5060;
    tag=1119358673
From: sip:control@192.168.12.158:5564;tag=1
Call-ID: 1119359020
CSeq: 1 INVITE
Contact: sip:192.168.12.150:5060
Content-Type: multipart/mixed;boundary=snow-boundary-1
Session-Expires: 6000
Content-Length: 468

--snow-boundary-1
Content-Type: application/mediaservercontrol+xml

<?xml version="1.0"?>
<MediaServerControl version="1.0">
    <response request="configure_leg" code="200"
        text="OK"/>
</MediaServerControl>
--snow-boundary-1
Content-Type: application/sdp
v=0
o=SnowShoreUaV1 294702567 1726956429
    IN IP4 192.168.12.150
s=SnowShore Sdp
t=0 0
m=audio 6010 RTP/AVP 0
c=IN IP4 192.168.12.150
a=sendrecv

```

```
a=ptime:20
a=rtpmap:0 PCMU/8000
```

```
--snow-boundary-1--
-----
```

- 4 Supervisor makes teammate of agent and can start coach conference with INFO message

```
INFO sip:192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5564;tag=1
To: sip:conf=1234@192.168.12.150:5060;
    tag=1119358673
Call-ID: 1119359020
Cseq: 2 INFO
Via: SIP/2.0/UDP 192.168.12.158:5564
Content-Type: application/mediaservercontrol+xml
Content-Length: 190
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_team action="add">
      <teammate id="agent" />
    </configure_team>
  </request>
</MediaServerControl>
```

Response from IP Media Server:

```
INFO sip:control@192.168.12.158:5564 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.150:5060
To: sip:control@192.168.12.158:5564;tag=1
From: sip:conf=1234@192.168.12.150:5060;
    tag=1119358673
Call-ID: 1119359020
User-Agent: SnowShoreMediaServer/1.4.0-050614A
CSeq: 2 INFO
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 231

<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_team" code="200" text="OK">
    <team id="super" numteam="1">
      <teammate id="agent">
        </teammate>
      </team>
    </response>
</MediaServerControl>
-----
```

## 5 Supervisor goes directly into coached conferencing in one step using the INVITE

```
INVITE sip:conf=1234@192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5564;tag=1
To: sip:conf=1234@192.168.12.150:5060
Subject: Call from Control
Cseq: 1 INVITE
Call-ID: 1119359610
Contact: <sip:control@192.168.12.158:5564>
Content-Type: multipart/mixed;boundary=salem
Via: SIP/2.0/UDP 192.168.12.158:5564
--salem
Content-Type: application/sdp
```

```
v=0
o=mserv 1 1119359610 IN IP4 192.168.12.151
s=Riptide_Media_Server
c=IN IP4 192.168.12.157
t=0 0
m=audio 32128 RTP/AVP 0
--salem
Content-Type: application/mediaservercontrol+xml
```

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_leg id="super" mixmode="private">
      <configure_team action="set">
        <teammate id="agent"/>
      </configure_team>
    </configure_leg>
  </request>
</MediaServerControl>

--salem--
```

### Response from IP Media Server:

```
SIP/2.0 200 Ok
Server: SnowShoreMediaServer/1.4.0-050614A
Via: SIP/2.0/UDP 192.168.12.158:5564
To: sip:conf=1234@192.168.12.150:5060;
   tag=1119359262
From: sip:control@192.168.12.158:5564;tag=1
Call-ID: 1119359610
CSeq: 1 INVITE
Contact: sip:192.168.12.150:5060

Content-Type: multipart/mixed;boundary=snow-boundary-1
Session-Expires: 6000
Content-Length: 549
```

```
--snow-boundary-1
Content-Type: application/mediaservercontrol+xml

<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_leg" code="200" text="OK"/>
    <team id="super" numteam="1">
      <teammate id="agent">
        </teammate>
      </team>
    </MediaServerControl>
--snow-boundary-1
Content-Type: application/sdp

v=0
o=SnowShoreUaV1 861021530 278722862 IN IP4
  192.168.12.150
s=SnowShore Sdp
t=0 0
m=audio 6012 RTP/AVP 0
c=IN IP4 192.168.12.150
a=sendrecv
a=ptime:20
a=rtpmap:0 PCMU/8000
--snow-boundary-1
-----
```

**6** A query on team list from the supervisor's leg

```
INFO sip:192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5564;
  tag=1
To: sip:conf=1234@192.168.12.150:5060;
  tag=1119359262
Call-ID: 1119359610
Cseq: 2 INFO
Via: SIP/2.0/UDP 192.168.12.158:5564
Content-Type: application/mediaservercontrol+xml
Content-Length: 167

<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_team action="query">
      </configure_team>
    </request>
  </MediaServerControl>
```

Response from IP Media Server:

```
INFO sip:control@192.168.12.158:5564 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.150:5060
```

```

To: sip:control@192.168.12.158:5564;tag=1
From: sip:conf=1234@192.168.12.150:5060;
    tag=1119359262
Call-ID: 1119359610
User-Agent: SnowShoreMediaServer/
    1.4.0-050614A
CSeq: 2 INFO
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 231

```

```

<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_team" code="200"
    text="OK">
    <team id="super" numteam="1">
      <teammate id="agent">
        </teammate>
      </team>
    </response>
  </MediaServerControl>

```

-----

## 7 Supervisor clears entire team list.

```

INFO sip:192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5564;tag=1
To: sip:conf=1234@192.168.12.150:5060;
    tag=1119359262
Call-ID: 1119359610
Cseq: 3 INFO
Via: SIP/2.0/UDP 192.168.12.158:5564
Content-Type: application/mediaservercontrol+xml
Content-Length: 147

```

```

<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_team action="set"/>
  </request>
</MediaServerControl>

```

### Response from the IP Media Server:

```

INFO sip:control@192.168.12.158:5564 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.150:5060
To: sip:control@192.168.12.158:5564;tag=1

From: sip:conf=1234@192.168.12.150:5060;tag=1119359262
Call-ID: 1119359610
User-Agent: SnowShoreMediaServer/1.4.0-050614A
CSeq: 3 INFO

```

```
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 191
```

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_team" code="200"
    text="OK">
    <team id="super" numteam="0">
    </team>
  </response>
</MediaServerControl>
```

-----

**8 Agent queries who is on the team**

```
INFO sip:192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5562;
  tag=1
To: sip:conf=1234@192.168.12.150:5060;
  tag=1119358479
Call-ID: 1119358826
Cseq: 2 INFO
Via: SIP/2.0/UDP 192.168.12.158:5562
Content-Type: application/mediaservercontrol+xml
Content-Length: 167
```

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <configure_team action="query">
    </configure_team>
  </request>
</MediaServerControl>
```

**Response from IP Media Server:**

```
INFO sip:control@192.168.12.158:5562 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.150:5060
To: sip:control@192.168.12.158:5562;tag=1
From: sip:conf=1234@192.168.12.150:5060;
  tag=1119358479
Call-ID: 1119358826
User-Agent: SnowShoreMediaServer/1.4.0-050614A
CSeq: 2 INFO
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 191
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="configure_team" code="200"
    text="OK">
```



```

        <team id="agent" numteam="0">
        </team>
    </response>
</MediaServerControl>

```

-----

## 9 Supervisor is added to the conference: mixmode=full

```

INFO sip:192.168.12.150:5060 SIP/2.0
From: sip:control@192.168.12.158:5564;tag=1
To: sip:conf=1234@192.168.12.150:5060;
    tag=1119359262
Call-ID: 1119359610
Cseq: 4 INFO
Via: SIP/2.0/UDP 192.168.12.158:5564
Content-Type: application/mediaservercontrol+xml
Content-Length: 148

```

```

<?xml version="1.0"?>
<MediaServerControl version="1.0">
    <request>
        <configure_leg mixmode="full"/>
    </request>
</MediaServerControl>

```

### Response from IP Media Server:

```

INFO sip:control@192.168.12.158:5564 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.150:5060
To: sip:control@192.168.12.158:5564;tag=1
From: sip:conf=1234@192.168.12.150:5060;
    tag=1119359262
Call-ID: 1119359610
User-Agent: SnowShoreMediaServer/1.4.0-050614A
CSeq: 4 INFO
Content-Type: application/mediaservercontrol+xml
Supported: timer
Supported: 100rel
Content-Length: 137

```

```

<?xml version="1.0"?>
<MediaServerControl version="1.0">
    <response request="configure_leg" code="200"
        text="OK" />
</MediaServerControl>

```

## IVR with MSCML

### Playing a Simple Announcement

#### Play Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <play id="332985001"
      prompt="file:///opt/snowshore/prompts/generic/
      10.ulaw"/>
  </request>
</MediaServerControl>
```

#### Expected Response

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="play" id="332985001"
    code="200" text="OK" reason="EOF"/>
</MediaServerControl>
```

### Playing a Sequenced Announcement

#### Play Payload

Example 39. Playing Sequenced Announcement—Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
<request>
<playcollect id=21212 maxdigits=5 firstdigittimer=250
  interdigittimer=100 extradigittimer=250>
<prompt baseurl="file:///opt/snowshore/
|prompts/generic/" locale="en_US">
  <audio url="num_changed.wav"/>
  <delay duration="10"/>
  <audio url="new_number.wav"/>
  <variable type="dig" subtype="ndn"
  value="9725551212"/>
  <audio url="make_note.wav"/>
</prompt>
</playcollect>
</request>
</MediaServerControl>
```

---

## Stopping a Play Command

### Request 1 Payload

#### Example 40. Request 1 Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <play id="332985011"
      prompt="file:///opt/snowshore/
      prompts/generic/
      10.ulaw"/>
    </request>
  </MediaServerControl>
```

### Request 2 Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <stop id="332985012" />
  </request>
</MediaServerControl>
```

### Expected Response to Request 1 Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="play" id="332985011"
    code="200" playduration="10000" text="OK"
    reason="stopped"/>
</MediaServerControl>
```

### Expected Response to Request 2 Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="stop" id="332985012"
    code="200" text="OK"/>
</MediaServerControl>
```

## PlayCollect

### Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <playcollect id="332986004"
      prompt="file:///opt/snowshore/prompts/generic/
      10.ulaw" maxdigits="1" barge="yes"/>
  </request>
</MediaServerControl>
```

### Expected Response

The play is barged into on the digit press. The operation matches the digit 2 and the INFO response indicates the match.

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="playcollect" id="332986004"
    code="200" text="OK" reason="match"
    playduration="10000" digits="2"/>
</MediaServerControl>
```

## Playing a Recording

### Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <playrecord id="332987015"
      prompt="file:///opt/snowshore/prompts/generic/en_U
      S/10.ulaw"
      recur="file:///var/snowshore/rec/temp01.ulaw"
      recencoding="ulaw"/>
  </request>
</MediaServerControl>
```

### Expected Response

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="playrecord" id="332987015"
    code="200" text="OK" reason="end_silence"
    reclength="1234" playduration="10000" digits="" />
</MediaServerControl>
```

## Stopping a Recording

### Request 1 Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <playrecord id="332987022"
      prompt="file:///opt/snowshore/prompts/generic/
      10.ulaw" initsilence="30000"
      recur1=file:///tmp/7001.ulaw"/>
  </request>
</MediaServerControl>
```

### Request 2 Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <request>
    <stop />
  </request>
</MediaServerControl>
```

### Expected Response Request 1 Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="playrecord" id="332987022"
    code="200" text="OK" reason="stopped" reclength="0"
    playduration="10000" digits="" />
</MediaServerControl>
```

### Expected Response Request 2 Payload

```
<?xml version="1.0"?>
<MediaServerControl version="1.0">
  <response request="stop" code="200" text="OK"/>
</MediaServerControl>
```

## Asynchronous DTMF

### Subscribing to Standard Digit Events

```
<?xml version="1.0"?>
  <MediaServerControl version="1.0">
    <request>
      <configure_leg>
        <subscribe>
          <events>
            <keypress report="standard"/>
          </events>
        </subscribe>
      </configure_leg>
    </request>
  </MediaServerControl>
```

### Subscribing to Long Digit Events

```
<?xml version="1.0"?>
  <MediaServerControl version="1.0">
    <request>
      <configure_leg>
        <subscribe>
          <events>
            <keypress report="long"/>
          </events>
        </subscribe>
      </configure_leg>
    </request>
  </MediaServerControl>
```

### Subscribing to Both Standard and Long Digit Events

```
<?xml version="1.0"?>
  <MediaServerControl version="1.0">
    <request>
      <configure_leg>
        <subscribe>
          <events>
            <keypress report="both"/>
          </events>
        </subscribe>
      </configure_leg>
    </request>
  </MediaServerControl>
```

## Turning Off Digit Event Reporting

```
<?xml version="1.0"?>
  <MediaServerControl version="1.0">
    <request>
      <configure_leg>
        <subscribe>
          <events>
            <keypress report="none"/>
          </events>
        </subscribe>
      </configure_leg>
    </request>
  </MediaServerControl>
```

## Example Responses

The following shows the response generated by detection of a standard "4" DTMF digit.

---

**Note:** This is the first digit detected so the interdigittime attribute has a value of 0.

---

```
<?xml version="1.0"?>
  <MediaServerControl version="1.0">
    <notification>
      <keypress digit="4" length="standard"
method="standard"
interdigittime="0">
        <status command="play" duration="10"/>
      </keypress>
    </notification>
  </MediaServerControl>
```

The following shows the response generated by detection of a long pound (#).

```
<?xml version="1.0"?>
  <MediaServerControl version="1.0">
    <notification>
      <keypress digit="#" length="long" method="long"
interdigittime="200">
        <status command="idle" duration="4"/>
      </keypress>
    </notification>
  </MediaServerControl>
```

[Example 41](#) shows the responses that are generated by the detection of two standard pound (#) events which also meet the double sequence criteria for long digits. In this scenario, three responses are generated; two for each of the standard pound events and one indicating that these responses comprise a long digit.

Multiple responses are generated because the subscription contained `<keypress report="both"/>`, as shown in [Example 41](#).

### Example 41. Asynchronous DTMF Response

```
1. <?xml version="1.0"?>
  <MediaServerControl version="1.0">
    <notification>
      <keypress digit="#" length="standard"
        method="standard" interdigittime="0">
        <status command="idle" duration="5"/>
      </keypress>
    </notification>
  </MediaServerControl>

2. <?xml version="1.0"?>
  <MediaServerControl version="1.0">
    <notification>
      <keypress digit="#" length="standard"
        method="standard"
        interdigittime="1000">
        <status command="idle" duration="6"/>
      </keypress>
    </notification>
  </MediaServerControl>

3. <?xml version="1.0"?>

  <MediaServerControl version="1.0">
    <notification>
      <keypress digit="#" length="long"
        method="double"
        interdigittime="1000">
      <status command="idle" duration="6"/>
      </keypress>
    </notification>
  </MediaServerControl>
```

If asynchronous DTMF is used in a conferencing scenario, make sure to maintain the subscription across multiple `<configure_leg/>` requests.



## Call Flow for IVR with MSCML

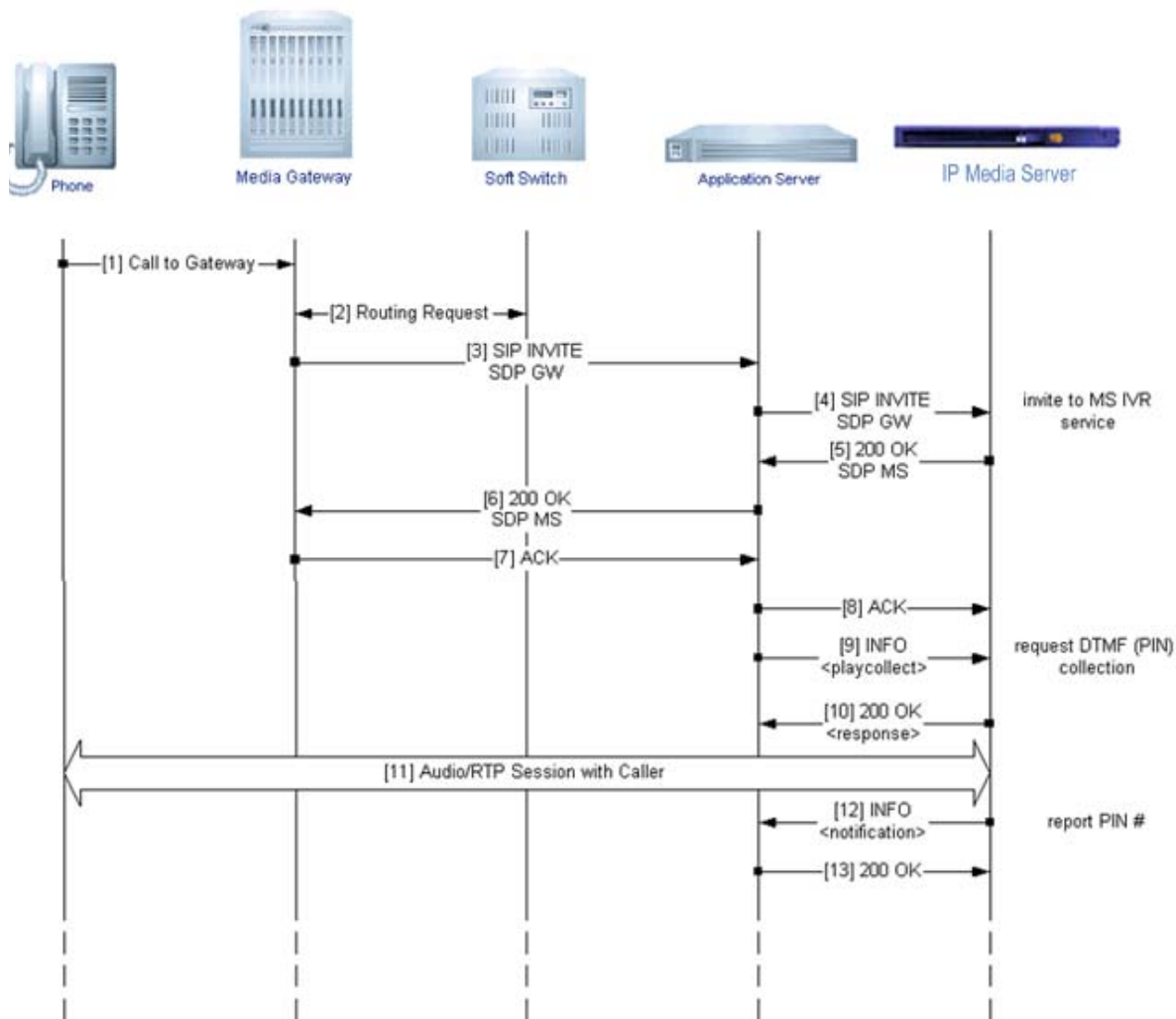


Figure 13. Call Flow: IVR with MSCML

## Call Flow for PIN Collection, IVR with MSCML

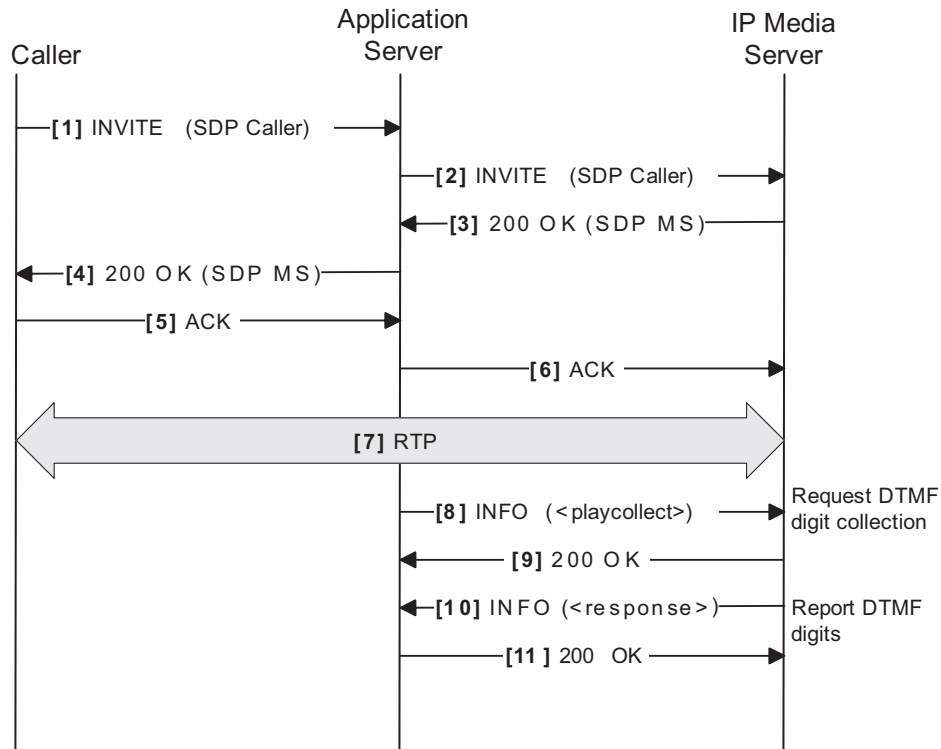


Figure 14. Call Flow: PIN Collection, IVR with MSCML

### Explanation of Call Flow

The numbers in the following table correspond to those on the call flow diagram in [Figure 14](#).

[1]	SIP INVITE from caller to application server Method: SIP INVITE Body: Caller's SDP
[2]	SIP INVITE from application server to IP Media Server Method: SIP INVITE Body: Caller's SDP
[3]	Response to SIP INVITE [2] Method: SIP INVITE Body: Dialogic® IP Media Server's SDP
[4]	Response to SIP INVITE [1] Method: SIP INVITE Body: Dialogic® IP Media Server's SDP
[5]	Acknowledgement of final response to INVITE [1] Method: SIP ACK Body: None
[6]	Acknowledgement of final response to INVITE [2] Method: SIP ACK Body: None

[7]	Bi-directional SDP established between caller and IP Media Server
[8]	<p>Application server directs Dialogic® IP Media Server to play a prompt and collect returned digits</p> <p>Method: SIP INFO</p> <p>Body:</p> <pre>&lt;?xml version="1.0"?&gt; &lt;MediaServerControl version="1.0"&gt;     &lt;request&gt; &lt;playcollect prompt="file:///opt/snowshore/prompts/conf/enterpin.wav" maxdigits="4" cleardigits="yes" returnkey="#" escapekey="*"&gt; &lt;/playcollect&gt;     &lt;/request&gt; &lt;/MediaServerControl&gt;</pre>
[9]	<p>Response from Dialogic® IP Media Server to &lt;playcollect&gt; request</p> <p>Method: SIP INFO</p> <p>Body: Empty</p>
[10]	<p>Return collected DTMF digits</p> <p>Method: SIP INFO</p> <p>Body:</p> <pre>&lt;?xml version="1.0"?&gt; &lt;MediaServerControl version="1.0"&gt; &lt;response request="playcollect" code="200" text="OK" reason="match" digits="1234" playduration="5230"/&gt; &lt;/MediaServerControl&gt;</pre>
[11]	<p>Response from application server to INFO response</p> <p>Method: SIP INFO</p> <p>Body: Empty</p>

## Call Flow for Recording a Message, IVR with MSCML

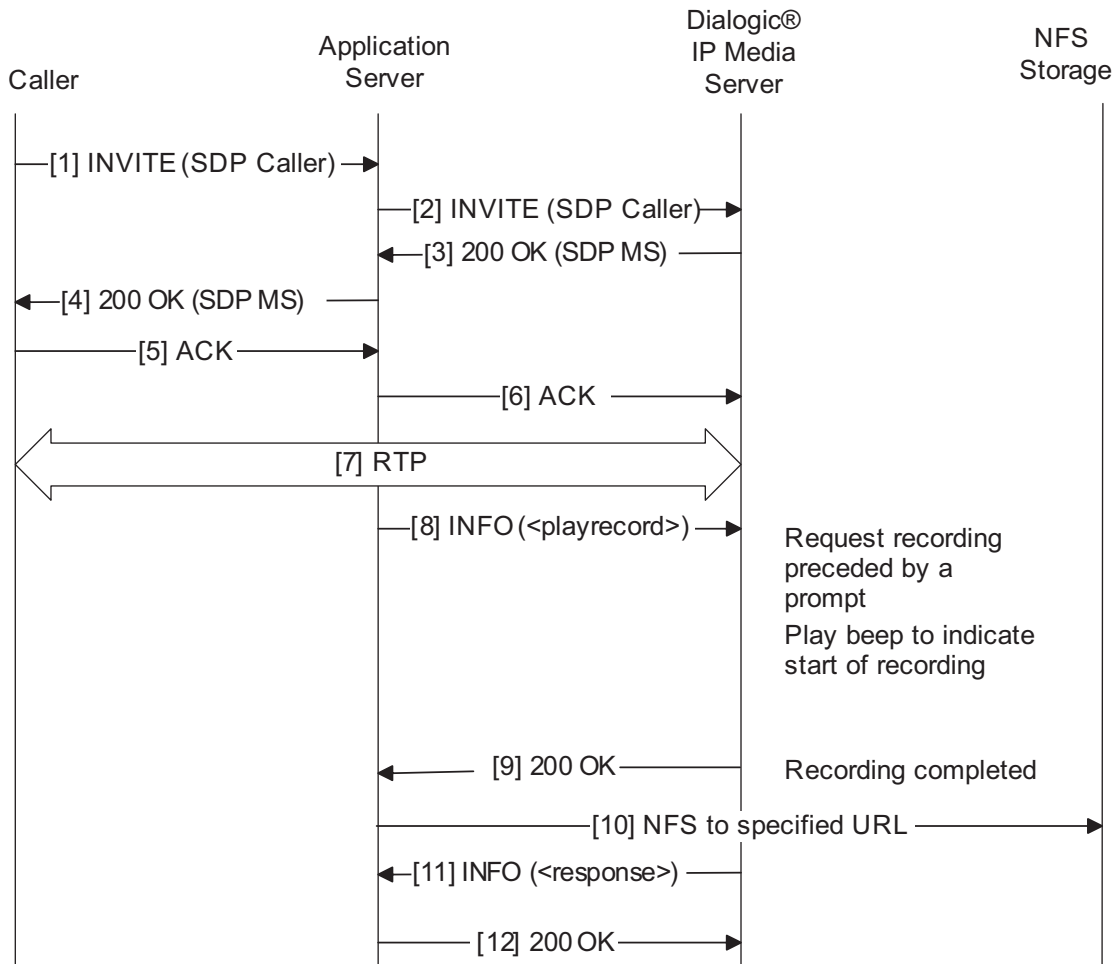


Figure 15. Call Flow: Recording, IVR with MSCML

## Explanation of Call Flow

The numbers in the following table correspond to those on the call flow diagram in [Figure 15](#).

[1]	SIP INVITE from caller to application server Method: SIP INVITE Body: Caller's SDP
[2]	SIP INVITE from application server to IP Media Server Method: SIP INVITE Body: Caller's SDP
[3]	Response to SIP INVITE [2] Method: SIP INVITE Body: Dialogic® IP Media Server's SDP
[4]	Response to SIP INVITE [1] Method: SIP INVITE Body: Dialogic® IP Media Server's SDP
[5]	Acknowledgement of final response to INVITE [1] Method: SIP ACK Body: None
[6]	Acknowledgement of final response to INVITE [2] Method: SIP ACK Body: None
[7]	Bi-directional SDP established between caller and IP Media Server
[8]	Application server directs IP Media Server to play a prompt and then record the caller's message. Method: SIP INFO Body: <pre>&lt;?xml version="1.0"?&gt; &lt;MediaServerControl version="1.0"&gt; request&gt; &lt;playrecord prompt="file:///opt/snowshore/prompts/conf/ recmessage.wav" escapekey="*" recurl="file:///storage/msg123.ulaw" initsilence="1000" endsilence="4000" beep="yes" recstopmask="01234356789*#"/&gt; &lt;/request&gt; &lt;/MediaServerControl&gt;</pre>
[9]	Response from Dialogic® IP Media Server to <playrecord> request Method: SIP INFO Body: Empty Response from Dialogic® IP Media Server to <playrecord> request

[10]	Recording is written to the external storage location specified in the url parameter of the <record> request.
[11]	Response from IP Media Server to application server when recording is complete Method: SIP INFO Body: <MediaServerControl version="1.0"> <response request="playrecord" code="200" text="OK" reason="endsilence" digits="" playduration="4738" reclength="24000"/> </MediaServerControl>
[12]	Response from application server to INFO response Method: SIP INFO Body: Empty

---

# VoiceXML

## Playing an Announcement

The sample code below is for a simple announcement, such as "The number you have reached...". This could be a dynamically generated script, which would fill in the values for the variables "was" and "is".

Example 42. Playing an Announcement, VoiceXML

```
<?xml version="1.0" ?>
<vxml version="1.0" application="default.xml">
  <form id="changed">
    <block>
      <var name="was" expr="9785551212"/>
      <var name="is" expr="9785554141"/>
      <prompt>
        <audio src="audio/numreached.ulaw"/>
        <value class="phone" mode="recorded" expr="was"/>
        <break/>
        <audio src="audio/beenchanged.ulaw"/>
        <audio src="audio/newnumberis.ulaw"/>

        <value class="phone" mode="recorded" expr="is"/>

        <audio src="audio/pleasenote.ulaw"/>
      </prompt>
    </block>
  </form>
</vxml>
```

## PIN Collection

This script collects a PIN number, and submits it to a web server for verification.

Example 43. PIN Collection

```
<?xml version="1.0" ?>

<vxml version="1.0" application="default.xml">

  <form id="get_pin">

    <field name="pin" type="digits?length=4">
      <prompt count="1">
```

```
<audio src="audio/enterpin.ulaw"> Please enter your PIN
number.</audio>
</prompt>
<prompt count="2">
  <audio src="audio/pin.ulaw"> PIN? </audio>

</prompt>
  <noinput count="1"> <reprompt/> </noinput>
  <noinput count="2"> <exit/> </noinput>
</field>

</field>
</form>
</vxml>
<field name="confirm" type="boolean">
  <prompt>
    <audio src="audio/yourpinis.ulaw">
Your pin is </audio>
    <value expr="pin"/>
    <audio src="audio/correct.ulaw">
      If this correct press 1. Otherwise, press 2
    </audio>
  </prompt>
  <nomatch count="1"> <reprompt/> </nomatch>
<nomatch count="3"> <exit/> </nomatch>
  <noinput count="1"> <reprompt/> </noinput>
  <noinput count="3"> <exit/> </noinput>
  <filled>
    <if cond="confirm">
      <submit namelist="pin" next="pin.cgi"/>
    <else/>
      <clear namelist="pin confirm"/>
    </if>
  </filled>
```



## Call Flow for VoiceXML

The following illustrates a call flow diagram for VoiceXML.

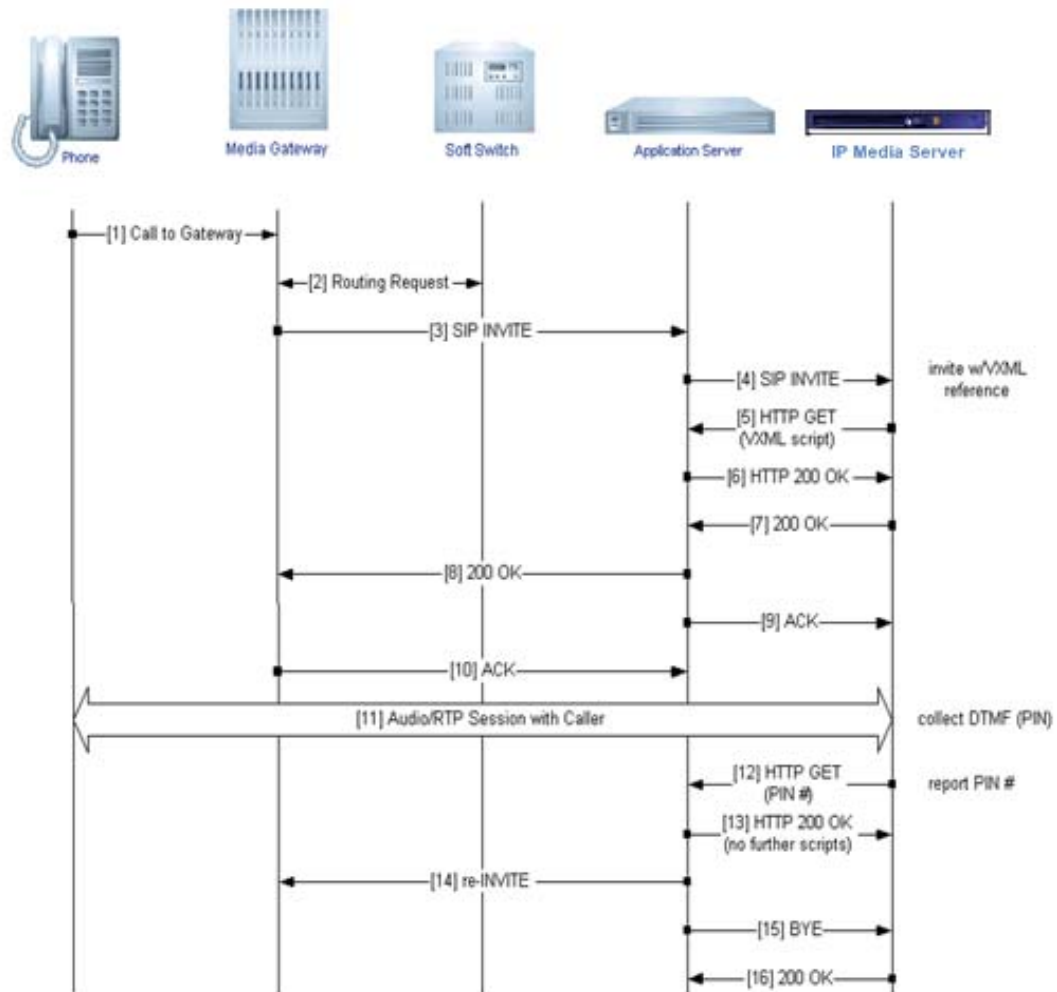


Figure 16. Call Flow: VoiceXML

## Transferring a Call

This script depicts a bridged transfer, which returns the caller to the original session with the interpreter when completed. One MSCML session (in addition to a VoiceXML resource) is required for each simultaneous transfer session. MSCML and VXML session configuration is performed at the CLI logical-vhost level.

### Example 44. Transferring a Call, VoiceXML

```
<?xml version="1.0"?>

<!-- 10.10.10.111 is the SnowShore MPS          -->
<!-- 12345679@10.10.10.253 is a valid SIP destination  -->
<!-- This script would first play "star", followed by "One" -->
<!-- Then attempts to do the transfer. If the transfer is -->
<!-- success, when the call is ended, you'll hear the reason-->
<!-- number (see below in the script) followed by the number-->
<!-- seconds the call is connected between the caller and -->
<!-- the callee.                                     -->
<vxml version="1.0">
<form name="genericTransfer">
<block>
<audio src="http://10.10.10.111/snowshore/prompts/gmvoices/star.neu" />
  </block>

<transfer name="makethecall" dest="sip:12345679@10.10.10.253" connecttimeout="30s"
  bridge="true">
  <prompt>
    <audio src="http://10.10.10.111/snowshore/prompts/gmvoices/1.ris" />
  </prompt>
  <filled>
    <if cond="makethecall == 'busy'">
    <prompt>

<audiosrc="http://10.10.10.111/snowshore/prompts/gmvoices/2.ris" />
  </prompt>
  <elseif cond="makethecall == 'noanswer'"/>
  <prompt>

<audio src="http://10.10.10.111/snowshore/prompts/gmvoices/3.ris" />
  </prompt>
  <elseif cond="makethecall == 'near_end_disconnect'"/>
  <prompt>
  <audio src="http://10.10.10.111/snowshore/prompts/gmvoices/4.ris" />
  </prompt>
  <elseif cond="makethecall == 'far_end_disconnect'"/>
  <prompt>

<audio src="http://10.10.10.111/snowshore/prompts/gmvoices/5.ris" />
  </prompt>
  <elseif cond="makethecall == 'network_disconnect'"/>
  <prompt>
```

---

```
<audio src="http://10.10.10.111/snowshore/prompts/gmvoices/6.ris" />
</prompt>

<elseif cond="makethecall == 'network_busy'"/>
  <prompt>
<audio src="http://10.10.10.111/snowshore/prompts/gmvoices/7.ris" />
</prompt>
<elseif />
<prompt>
<value class="number" expr="makethecall$.duration" />
<audio src="http://10.10.10.111/snowshore/prompts/gmvoices/seconds.neu" />
</prompt>
</if>
</filled>
</transfer>
<block>
  <audio src="http://10.10.10.111/snowshore/prompts/gmvoices/star.neu" />
</block>
</form>
</vxml>
```

## Using VCR and Gain Controls

The following properties are used for VCR and gain controls in the VoiceXML 2.0 browser:

- ◆ `com.snowshore.vcr_skip`: This property contains the amount of time to skip ahead or behind when one of the 'vcr\_fwd' or 'vcr\_rew' grammars are matched. This value is a Time Designation type defined in section 6.5 of the VoiceXML 2.0 specification and has a default value of '5s' if the property is not set.
- ◆ `com.snowshore.gain_adj`: This property contains the amount of gain to adjust the audio playback when 'gain\_up' or 'gain\_down' grammars are matched. This value is in dB and has a default value of '3' if the property is not set.
- ◆ `com.snowshore.vcr_pause`: This property contains the amount of time to pause a prompt when the 'vcr\_pause' grammar is matched. This value is a Time Designation type defined in section 6.5 of the VoiceXML 2.0 specification and has a default value of '3s' if the property is not set.
- ◆ `com.snowshore.vcr_speed_adj`: This property contains the percentage to adjust the playback speed of the audio prompts when one of the 'vcr\_faster' or 'vcr\_slower' grammars are matched. This value is a percentage.

For an example script, see [“Using VCR and Gain Controls in VoiceXML 2.0” \(page 301\)](#).

---

**Note:** VCR controls for video files are supported through the Real Time Streaming Protocol (RTSP) only.

---

Table 49 describes the VCR and gain controls that are supported through the RTSP in VoiceXML 2.0. For an example script, see [“Using VCR Controls for RTSP in VoiceXML 2.0” \(page 302\)](#).

**Table 49.** VCR and Gain Controls Support for RTSP in VoiceXML 2.0

Property	Description	RTSP Support
<code>com.snowshore.vcr_skip</code>	fast forward and rewind	Yes
<code>com.snowshore.gain_adj</code>	gain adjustment	No
<code>com.snowshore.vcr_pause</code>	pause	Yes
<code>com.snowshore.speed_adj</code>	speed adjustment	No

The VoiceXML 2.0 browser supports built-in grammars to support VCR and gain controls:

```
<grammar mode"dtmf" src="builtin:dtmf/vcr"/>
<grammar mode"dtmf" src="builtin:dtmf/gain"/>
```

The following is an example script using VCR and gain control in VoiceXML 2.0.

#### Example 45. Using VCR and Gain Controls in VoiceXML 2.0

```
<?xml version="1.0"?>
<vxml version="2.0" xmlns="http://www.w3.org/2001/vxml">

<property name="com.snowshore.gain_adj" value="10" />
<property name="com.snowshore.vcr_skip" value="5s" />
<property name="com.snowshore.vcr_pause" value="3s" />
<property name="com.snowshore.vcr_speed_adj" value="5" />

<form id="g">
  <field name="u">
    <link event="modify_audio_control">
      <grammar src="builtin:dtmf/gain?down=7;up=8;reset=9"/>
    </link>

    <link event="modify_audio_control">
      <grammar src="builtin:dtmf/vcr?rew=1;fwd=2;pause=3;slower=4;faster=5"/>
    </link>

    <prompt bargein="true" vcr="true">
      <audio src="file:///opt/showshore/prompts/generic/en_US/num_changed.ulaw"/>
      <audio src="http://localhost/showshore/prompts/number.wav"/>
      <audio src="http://localhost/showshore/prompts/alpha.wav"/>
    </prompt>

    <catch event="modify_audio_control">
      <clear/>
      <reprompt/>
    </catch>

  </field>
</form>
</vxml>
```

The following is an example script using VCR controls for the RTSP in VoiceXML 2.0.

### Example 46. Using VCR Controls for RTSP in VoiceXML 2.0

```
<?xml version="1.0"?>
<vxml version="2.0" xmlns="http://www.w3.org/2001/vxml">

<property name='com.snowshore.vcr_skip' value='6s' />
<property name='com.snowshore.vcr_pause' value='5s' />

<form id="g">
  <field type="boolean" name="u">
    <link event="modify_video_control">
      <grammar src="builtin:dtmf/vcr?rew=0;fwd=6;pause=3" />
    </link>

    <prompt bargein='true' vcr='true'>
      <audio src="rtsp://rtspserver.dialogic.com/sample.3gp" />
    </prompt>

    <catch event="modify_video_control">
      <clear />
      <reprompt />
    </catch>
  </field>
</form>
</vxml>
```

The following is an example script using VCR controls in VoiceXML 1.0.

### Example 47. Using VCR Controls in VoiceXML 1.0

```

<link event="vcr">
  <dtmf>
    7 {vcr_rew} | 8 {vcr_pause} | 9 {vcr_fwd}
  </dtmf>
</link>

<form id="form1">
<!-- Remove the default termchar, as we don't want to wait for it -->
  <property name="termchar" value=""/>
  <property name="com.snowshore.vcr_skip" value="1s"/> <!-- default 5s -->
  <property name="com.snowshore.vcr_pause" value="10s"/> <!-- default 10s -->
<!-- VCR controls work on this prompt only -->
<block>
  <prompt vcr="true">
<audio
src="file:///net/192.168.12.12/opt/snowshore/prompts/gmvoices/
  en_US/susan/msg055"/>

<!-- The message audio -->audio
src="file:///net/192.168.12.12/opt/snowshore/prompts/generic/
  1MinuteMusic.raw"/>
</prompt></prompt>
</block>
<!-- This is the menu "To repeat, press 1, to save, press 2, to erase ... -->
<!-- VCR controls don't affect this prompt, but ARE active, i.e. 7 can -->
<!-- backup out of this menu back into the message -->
<field name="field1" type="digits?length=1">
  <prompt>
audio
src="file:///net/192.168.12.12/opt/snowshore/prompts/generic/
  en_US/try_again.ulaw"/>
  </prompt>
filled>

  <!-- Fetch next thing to do -->
<submit
next="msgfetch?sid=12345&msgid=42&digit=${field1}"/>
  </filled>

<catch event="vcr">
  <!-- VCR events must loop back to the top in order to be effective-->
  <clear/>
  <reprompt/>
</catch>

</field>
</form>

```

## Text Overlay

This section provides VoiceXML 2.0 text overlay sample code.

### Simple Text Overlay

This script illustrates a simple text overlay using VoiceXML 2.0. Modify the script, such as audio source file and font name, to suit your implementation.

```
<?xml version="1.0" encoding="UTF-8" ?>
<vxml version="2.0" xmlns="http://www.w3.org/2001/vxml">

<form id="form1">
  <block>
    <prompt bargein="false">
      <audio src="file:///opt/snowshore/htdocs/3GPP/sample1.3gp"/>
      <overlay position="x0y240" size="x346y20">
        <content>
          <font name="font:///opt/snowshore/fonts/FreeSans.ttf" size="14"
fgcolor="yellow"/>
            "Hello, world!"
          </content>
        </overlay>
      </prompt>
    </block>
  </form>
</vxml>
```

### Scrolling Text Overlay

This script illustrates a scrolling text overlay using VoiceXML 2.0. Modify the script, such as audio source file and font name, to suit your implementation.

```
<?xml version="1.0" encoding="UTF-8" ?>
<vxml version="2.0" xmlns="http://www.w3.org/2001/vxml">

<form id="form1">
  <block>
    <prompt bargein="false">
      <audio src="file:///opt/snowshore/htdocs/3GPP/sample2.3gp"/>
      <overlay position="x0y40" size="x346y20" border-width="2" border-color="yellow"
bgcolor="0xFFFF007F">
        <content>
          <font name="font:///opt/snowshore/fonts/FreeSans.ttf" size="14"
fgcolor="black"/>
            <scroll mode="continuous" speed="40" direction="left"/>
            "The quick brown fox jumps over the lazy dog."
          </content>
        </overlay>
      </prompt>
    </block>
  </form>
</vxml>
```



# A - Audio Library

This appendix describes a preconfigured sound library (audio files) which consists of phrases and messages, numbers, time and money, quantities, and miscellaneous words.

This appendix includes the following major sections:

- ◆ [Sound Library](#)
- ◆ [Generic Audio Files](#)

## Sound Library

The Dialogic® IP Media Server comes with a library of stock audio segments, professionally recorded in a female voice. These files are stored in the following location:

```
/opt/snowshore/prompts/
```

Within this directory are two subdirectories:

- ◆ /generic/ contains some files recorded by Dialogic that can be useful for testing.
- ◆ /gmvoices/ holds a directory of audio files in US English (/en\_US/susan/) and a directory of audio files in Chinese Mandarin (/zh\_CN/sara/).

These audio segments provide a consistent set of sounds that have been recorded to support a wide variety of pre-programmed phrases, spoken numbers, dates, and other sequenced recordings.

These files are in a protected area and cannot be modified. If you have application-specific audio files, Consider placing these files an NFS server on your network rather than locally on the IP Media Server.

With the exception of the generic audio files and audio files for letters of the alphabet, the other files listed in the following pages exist in both US English and Chinese Mandarin versions.

The tables on the following pages explain the categories of phrases provided with the prompt library. Phrases are categorized as follows:

- ◆ [Phrases and Messages](#)
- ◆ [Numbers](#)
- ◆ [Letters](#)
- ◆ [Time and Money](#)
- ◆ [Press Keys](#)
- ◆ [Quantities](#)
- ◆ [Miscellaneous Words](#)

### Phrases and Messages

Phrases and message are located in:

```
opt/snowshore/prompts/gmvoices/en_US/susan/
```

and in

```
/opt/snowshore/prompts/gmvoices/zh_CN/sara/
```

These files are standard phrase segments that can be combined into sequences.

Table 50. Prompt Library: Standard Phrases

Filename	Spoken as
msg001	one second of silence
msg002	600Hz beep tone
msg003	Goodbye.
msg004	Hello.
msg005	Good morning.
msg006	Good afternoon.
msg007	Good evening.
msg008	If this is correct...
msg009	To transfer to another extension...
msg010	To return to the main menu...
msg011	To return to the previous menu...
msg012	To delete this message...
msg013	To delete all messages...
msg014	To exit the system...
msg015	To end this call...
msg016	To cancel delivery of this message...
msg017	To send your message now...
msg018	Please hold while your call is being transferred.
msg019	Please leave your message after the tone.
msg020	Please begin recording after the tone.
msg021	Thank you for calling.
msg022	Thank you for calling, goodbye.
msg023	Thank you.
msg024	Please enter your mailbox number.
msg025	Please enter your passcode.
msg026	I'm sorry, that passcode is invalid. Please reenter your passcode.
msg027	I'm sorry, that mailbox number is invalid. Please reenter your mailbox number.

Table 50. Prompt Library: Standard Phrases (continued)

Filename	Spoken as
msg028	I'm sorry, that is an invalid entry. Please try again.
msg029	You have no messages.
msg030	You have no more messages.
msg031	You have...
msg032	...new message.
msg033	...new messages.
msg034	...saved message.
msg035	...saved messages.
msg036	Your mailbox is currently full.
msg037	Message deleted.
msg038	Message saved.
msg039	Please hold.
msg040	Please hold for assistance.
msg041	I'm sorry...
msg042	You entered...
msg043	End of messages.
msg044	We cannot identify that entry. Please try again.
msg045	Please hold while I transfer your call to the operator.
msg046	We're sorry you are having difficulty. Please try your call again later.
msg047	If this is correct, press 1. If not, press 2.
msg048	Please enter your passcode now.
msg049	If you're calling from a touchtone phone, press 1 now. Otherwise, please stay on the line to speak with an operator.
msg050	Please enter your business phone number beginning with the area code now.
msg051	You have selected...
msg052	You have reached...
msg053	To make another selection...
msg054	To try again...

Table 50. Prompt Library: Standard Phrases (continued)

Filename	Spoken as
msg055	Extension...
msg056	The telephone number you entered is...
msg057	The fax number you entered is...
msg058	Please enter the account number.
msg059	Please enter your personal identification number.
msg060	Your personal identification number is...
msg061	Thank you, please hold.
msg062	Please record your message at the tone. When you are finished you may hang up or...
msg063	...for more options.
msg064	Please record your greeting at the tone.
msg065	To save this message, press...
msg066	To delete this message, press...
msg067	To listen to the next message, press...
msg068	To replay this message, press...
msg069	The passcode you entered is...
msg070	Your message to...
msg071	...was delivered.
msg072	...could not be delivered.
msg073	I'm sorry, we cannot connect your call at this time. Please try again later.
msg074	Mailbox number...
msg075	...is full.

## Numbers

These files are spoken as cardinal numbers. (Ordinals are listed with dates.)

Some numbers have multiple files for neutral, rising, and falling inflection, as indicated by their file extensions.

All these files are located in:

```
opt/snowshore/prompts/gmvoices/en_US/susan/
```

and in

/opt/snowshore/prompts/gmvoices/zh\_CN/sara/

Table 51. Prompt Library: Cardinal Numbers

Filename	Numbers	Extension	Spoken As
0.neu ... through 9.neu 0.ris ... through 9.ris 0.dwn ...through 9.dwn	0 through 9	.neu .ris .dwn	Zero through Nine with neutral, rising, or falling intonation
10.num, 11.num, 12.num... through 100.num	10 through 100	.num	Ten, Eleven, Twelve,... One hundred'
100.neu ... through 1000.neu 100.ris ... through 1000.ris 100.dwn ...through 1000.dwn	100 through 1000 in increments of 100	.neu .ris .dwn	One hundred through One thousand with neutral, rising, or falling intonation.
thousand.num million.num billion.num trillion.num	thousand, million, billion, and trillion	.num	Thousand, Million, Billion, Trillion

## Dates and Ordinal Numbers

The following files consists of dates and ordinal numbers.

Table 52. Prompt Library: Dates and Ordinal Numbers

Filename	Days/Months	Spoken As
mon.dat tue.dat wed.dat thu.dat fri.dat sat.dat sun.dat	Monday through Sunday	Day of the week, Monday through Sunday

Table 52. Prompt Library: Dates and Ordinal Numbers (continued)

Filename	Days/Months	Spoken As
1st.dat, 2nd.dat, 3rd.dat... through 31st.dat	1st, 2nd, 3rd... through 31st	Spoken as ordinal number First, Second, Fourth, Thirty-first
jan.dat feb.dat mar.dat apr.dat may.dat jun.dat jul.dat aug.dat sep.dat oct.dat nov.dat dec.dat	January through December	Name of month January, February, etc.
Filename	Years	Spoken As
1975.dat ... through 2015.dat	1975 through 2015	Spoken as year. Nineteen seventy five, 'Two thousand and fifteen

## Letters

There are two audio files for each letter of the alphabet: one with rising intonation, for example `aup.ltr` and one with falling intonation, for example `adn.ltr`.

Having both types of files allows you to select audio files by context. For example, the letters C and N have different sounds in the acronyms for NBC and CNN.

These audio files are located in:

```
opt/snowshore/prompts/gmvoices/en_US/susan/
```

and in

```
/opt/snowshore/prompts/gmvoices/zh_CN/sara/
```

Filename	Spoken With	Example
aup through zup.ltr	Rising intonation	aup.ltr; bup.ltr ... through zup.ltr
adn through zdn.ltr	Falling intonation	adn.ltr; bdn.ltr ... through zdn.ltr

## Time and Money

You can combine these audio files with audio files for numbers to form sequences for times of day and money values. These audio files are located in:

```
opt/snowshore/prompts/gmvoices/en_US/susan/
```

and in

```
/opt/snowshore/prompts/gmvoices/zh_CN/sara/
```

Table 53. Prompt Library: Time and Money Phrases

Filename	Spoken as Time Zones:
est.tim	Eastern Standard Time
cst.tim	Central Standard Time
mst.tim	Mountain Standard Time
pst.tim	Pacific Standard Time
am.tim	AM
at.tim	at (time)
mid.tim	Midnight
noon.tim	Noon
oclock.tim	o'clock
pm.tim	PM
and.mon	and (money)
cent.mon	Cent
cents.mon	Cents
dollar.mon	Dollar
dollars.mon	Dollars



## Press Keys

Most of these files have two values: one with falling intonation (\*.dwn) and one with neutral intonation (\*.neu).

Press key audio files are located in:

```
opt/snowshore/prompts/gmvoices/en_US/susan/
```

and in

```
/opt/snowshore/prompts/gmvoices/zh_CN/sara/
```

Table 54. Prompt Library: Press Key Phrases

Filename	Spoken as Press Key Phrases:
press	Press
prs0.dwn / prs0.neu	Press 0
prs1.dwn / prs1.neu	Press 1
prs2.dwn / prs2.neu	Press 2
prs3.dwn / prs3.neu	Press 3
prs4.dwn / prs4.neu	Press 4
prs5.dwn / prs5.neu	Press 5
prs6.dwn / prs6.neu	Press 6
prs7.dwn / prs7.neu	Press 7
prs8.dwn / prs8.neu	Press 8
prs9.dwn / prs9.neu	Press 9
asterisk.dn / asterisk.neu	Asterisk
pound.dwn / pound.neu	Pound
star.dwn / star.neu	Star

## Quantities

Quantity audio files are located in:

`opt/snowshore/prompts/gmvoices/en_US/susan/`

and in

`/opt/snowshore/prompts/gmvoices/zh_CN/sara/`

Table 55. Prompt Library: Quantities

Filename	Spoken as	
	year.neu	year
years.neu	years	neutral
month.neu	month	neutral
months.neu	months	neutral
day.neu	day	neutral
days.neu	days	neutral
hour.neu	hour	neutral
hours.neu	hours	neutral
minute.neu	minute	neutral
minutes.neu	minutes	neutral
second.neu	second	neutral
seconds.neu	seconds	neutral
dollar.neu	dollar	neutral
dollars.neu	dollars	neutral
cent.neu	cent	neutral
cents.neu	cents	neutral
minus.dwn	minus	neutral
year.dwn	year	final intonation
years.dwn	years	final intonation
month.dwn	month	final intonation
months.dwn	months	final intonation
day.dwn	day	final intonation
days.dwn	days	final intonation

Table 55. Prompt Library: Quantities (continued)

Filename	Spoken as	
hour.dwn	hour	final intonation
hours.dwn	hours	final intonation
minute.dwn	minute	final intonation
minutes.dwn	minutes	final intonation
second.dwn	second	final intonation
seconds.dwn	seconds	final intonation
dollar.dwn	dollar	final intonation
dollars.dwn	dollars	final intonation
cent.dwn	cent	final intonation
cents.dwn	cents	final intonation
minus.dwn	minus	final intonation

## Miscellaneous Words

Miscellaneous words are located in:

`opt/snowshore/prompts/gmvoices/en_US/susan/`

and in

`/opt/snowshore/prompts/gmvolices/zh_CN/sara/`

---

**Note:** Many of these files do not have filename extensions.

---

Table 56. Prompt Library: Miscellaneous Words

Filename	Spoken as
act	Activate
acted	Activated
dash	Dash
deact	Deactivate
deacted	Deactivated
for	For
no	No

Table 56. Prompt Library: Miscellaneous Words (continued)

Filename	Spoken as
none	None
o.dwn o.neu o.ris	O
off.dwn off.neu	Off
on.dwn on.neu	On
or	Or
percent	Percent
point	Point
sent	Sent

## Generic Audio Files

Generic audio files are located in:

`/opt/snowshore/prompts/generic/en_US`

These files were internally recorded by Dialogic and can be useful for testing purposes.

**Table 57. Generic Prompt Phrases**

<b>Filename</b>	<b>Spoken as</b>
ac_changed.ulaw ac_changed.wav	"The area code for the number you are dialing has changed to..."
circuit_busy.ulaw circuit_busy.wav	"We're sorry. All circuits are busy now."
contact_provider.ulaw contact_provider.wav	"Please contact your service provider."
dial_again.ulaw dial_again.wav	"If you'd like to make a call, please hang up and dial again."
dial_operator.ulaw dial_operator.wav	"If you need help, please hang up and dial your operator."
disconnected.ulaw disconnected.wav	"...has been disconnected."
make_note.ulaw make_note.wav	"Please make a note of it."
new_number.ulaw new_number.wav	"The new number is..."
no_permission.ulaw no_permission.wav	"You no longer have permission to utilize the system."
num_changed.ulaw num_changed.wav	"The number you are calling has changed."
num_dialed.ulaw num_dialed.wav	"The number you have dialed..."
num_invalid.ulaw num_invalid.wav	"...is not a valid number."

Table 57. Generic Prompt Phrases (continued)

Filename	Spoken as (continued)
please_check.ulaw please_check.wav	"Please check the number and dial again."
service_outage.ulaw service_outage.wav	"Please contact your service provider for information relating to this service outage."
try_again.ulaw try_again.wav	"Please hang up and try your call again later."

# B - VoiceXML Version 1.0 and Dialog Service

This appendix explains the basics of VoiceXML (VXML) Version 1.0, lists the supported VoiceXML elements and attributes, and explains ECMAScript language functionality.

---

**Note:** For information about VoiceXML 2.0, see [“VoiceXML Version 2.0 and Dialog Service” \(page 182\)](#).

---

The dialog service is one of two interfaces that the IP Media Server offers for developing IVR/DTMF applications. The other service is ivr through which SIP requests are enhanced by MSCML message bodies for play, playcollect, and playrecord. For details on IVR, see [Chapter 5, “IVR with MSCML”](#).

This appendix includes the following sections:

- ◆ [VoiceXML Interpreter](#)
- ◆ [VoiceXML Launcher](#)
- ◆ [Dialog Service Indicator and Request URI](#)
- ◆ [VoiceXML Concepts](#)
- ◆ [VoiceXML Application and Its Documents](#)
- ◆ [Dialogs](#)
- ◆ [Grammar and Scripting](#)
- ◆ [Session Variables](#)
- ◆ [File Storage and Retrieval](#)
- ◆ [Media Content Recovery Extension](#)
- ◆ [VoiceXML Elements Reference](#)

- ◆ [VoiceXML Attributes Reference](#)
- ◆ [VoiceXML Properties](#)
- ◆ [ECMAScript Functionality](#)
- ◆ [Support for VoiceXML Extended Session Variables](#)



## About VoiceXML

VoiceXML is a W3C standard scripting language for playing audio prompts and for collecting DTMF input.

The interpreter executes VoiceXML dialogs on an RTP stream (see next section). Each dialog represents an announcement, menu, or other IVR script. The dialogs finish when they have posted information to a Web server or returned a namelist back to the command that invoked the browser. If additional dialogs are needed, then the application runs another script.

While the media server supports full ECMA (European Computer Manufacturer's Association) Scripting abilities, Dialogic recommends that you perform that type of logic on the application server rather than in the VoiceXML script. This method reduces the processing and memory required by the Dialogic® IP Media Server and allows the Dialogic® IP Media Server to handle a large volume of simultaneous sessions.

### VoiceXML Interpreter

The IP Media Server includes interpreters for VoiceXML 1.0 and VoiceXML 2.0.

---

**Note:** For information about VoiceXML 2.0, see [“VoiceXML Version 2.0 and Dialog Service” \(page 182\)](#).

---

If the IP Media Server receives a SIP INVITE request directed to the dialog service, a VoiceXML session begins. The VoiceXML session initially gets (fetches) and executes the VoiceXML script that is specified in the SIP Request-URI parameter, `voicexml`.

Once specified, a VoiceXML script URI remains in effect until it completes execution or the session is stopped.

### Dialog Service Indicator and Request URI

Through the dialog service (`dialog`), the Dialogic® IP Media Server executes VoiceXML documents to offer IVR scripting with DTMF input and recorded audio output.

The application references the initial VoiceXML script using the Request-URI parameter `voicexml`. The following SIP Request-URI directs the IP Media Server to retrieve and execute `script1.vxml` from the server `app1.carrier.com`.

```
INVITE sip:dialog@MS_IP;voicexml=http://
    appl.carrier.com/path/
    script1.vxml
```

It is common for the HTTP URI referencing the VoiceXML script to be a query that contains its own parameters so the script can be dynamically generated. For example, the following HTTP URI provides a subscriber ID so the VoiceXML script can be appropriately personalized.

```
INVITE sip:dialog@MS_IP;voicexml=http://
  appl.carrier.com/cgi/bin/
  genvxml.pl?subscriberid=34590087
```

If a query HTTP URI is used, be careful to replace characters that are reserved in SIP with their hexadecimal equivalents preceded by a percent (%) character. This substitution is called escaping. In particular, the question mark (?) and equals sign (=) must be escaped to conform with SIP standards. The following example shows the escaped form of the previous example.

```
INVITE sip:dialog@MS_IP;voicexml=http://
  appl.carrier.com/cgi/bin/
  genvxml.pl%3Fsubscriberid%3D34590087
```

The SIP URI must be escaped, as described in [“Syntax and Escaping” \(page 88\)](#).

## VoiceXML Launcher

If a SIP INVITE to the Dialogic® IP Media Server is directed to the dialog service but does not specify a file (voicexml=), then the Dialogic® IP Media Server launches the dialog service and runs a default script configured by the system administrator using the Web UI MEDIA SERVER > VOICEXML menu.

## VoiceXML Concepts

### Syntax

VoiceXML (VXML) is an eXtensible Markup Language for the creation of IVR and Automated Speech Recognition (ASR) applications. Based on XML tag/attribute format, its syntax involves enclosing instructions (items) within a tag structure. For example:

```
<element_nameattribute_name="attribute_value">
  ...contained items...
</element_name>
```

Generic XML concepts remain unchanged in VoiceXML. For example, any character data must be escaped as per the XML specification.

### Scope

As for all XML, VoiceXML observes a hierarchical structure and applies the construct of scope to define the range within the source where a variable, event handler, or other element is applicable.

## Resource Fetching

VoiceXML defines several attributes to qualify properties relevant to the caching and fetching of documents and other resources.

---

Note: The Dialogic® IP Media Server does not currently support these attributes.

---

## VoiceXML Application and Its Documents

A VoiceXML application is a set of VoiceXML scripts that share the same root document. The application consists of one or more text files called documents. Document files are identified by a .vxml extension and are retrieved using HTTP or (Network File System) NFS.

If a VoiceXML application includes multiple documents, one of these can be the application root document. The application shares the root document among all other documents. Sharing root documents occurs when the script author explicitly sets the root document to be the same in multiple VoiceXML scripts. The IP Media Server's VoiceXML browser supports the VoiceXML standard.

Whenever a user interacts with a document in an application, the corresponding root document is also loaded. The root document is unloaded when a user transitions to a document not in the application.

While the root document is loaded, the application root document's variables are available to other documents as application variables. Grammars defined in the root document also remain active for the duration of the application.

Figure 17 shows how a root document is shared in a VoiceXML application.

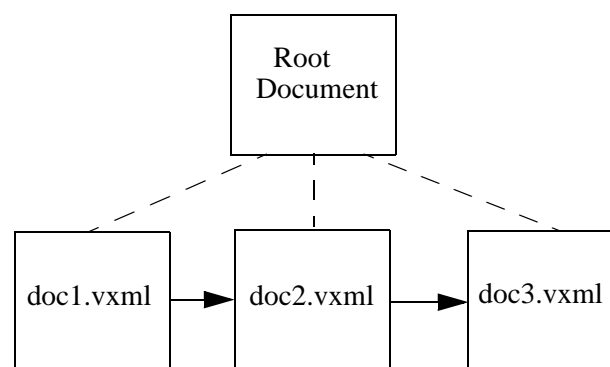


Figure 17. Sharing of Root Document in VoiceXML Application

## Dialogs

A VoiceXML document contains dialogs or conversations between a prompting VoiceXML interpreter and a responding caller.

Conversational building blocks are as follows:

- ◆ Prompts (in the Dialogic® IP Media Server context for playing audio)
- ◆ The grammar that specifies the range of acceptable user responses to a prompt
- ◆ Transitions (to the URI of the next prompt/response exchange)

Conversational constructs are as follows:

- ◆ Forms that collect an input or event, process it, and select the next form to visit.
- ◆ Menus for organization and navigation (multiple-choice options, with a transition for each).
- ◆ Links that execute a transaction or throw an event.

## Grammar and Scripting

The Dialogic® IP Media Server supports DTMF grammars and also supports ECMA tags that involve the scripting capabilities of the VoiceXML interpreter. See [“ECMAScript Functionality” \(page 361\)](#).

However, executing complex ECMAScript on the IP Media Server can affect its ability to perform real-time media processing. To conserve processing capacity and memory, Dialogic recommends limiting the use of ECMAScripting on the IP Media Server.

## Session Variables

Voice XML provides a facility, called session variables, to pass signaling information to Voice XML scripts. The IP Media Server supports the standard Voice XML session variables as well as extensions which capture additional information from the SIP INVITE. See [“Support for VoiceXML Extended Session Variables” \(page 364\)](#) for further details.

## File Storage and Retrieval

You can access VoiceXML scripts through the network using NFS and HTTP protocols. Audio and video content can be retrieved and stored using HTTP and NFS as well.

---

**Note:** The VoiceXML files can be located anywhere as long as the VoiceXML interpreter can access them.

---

## Media Content Recovery Extension

The Media Content Recovery mechanism has been extended such that there can be increased reliability of recorded content delivery from Voice XML applications.

"Next-generation" application architectures such as SIP and VoiceXML are highly distributed in nature and rely on multiple components and network communication protocols to function. In the case of VoiceXML applications, the link between the VoiceXML browser (the IP Media Server) and the application server, which provides the scripts and content, is critical.

Now that SIP and VoiceXML applications have been successfully deployed, carriers expect reliability and behavior similar to the legacy applications that are being replaced. In most situations, it is acceptable for a subscriber to call back and establish a new VoiceXML session if a failure occurs. The notable exception is when the caller has successfully recorded a message and expects for it to be sent. If the failure occurs before the message can be transmitted, there is no way to inform the user of the issue and there is nothing the user can do about it. The Media Content Recovery extension provides a solution to the specific issue of reliable content delivery.

Recording is implemented in VoiceXML through the `<record>` element. When a `<record>` element is processed, the IP Media Server creates a temporary file with a locally unique identifier name.

The IP Media Server VoiceXML 1.0 browser maintains a list of the temporary files created during each call and deletes them when the VoiceXML script terminates. The browser also deletes any temporary files upon startup. When the Media Content Recovery mechanism is used, temporary recordings that are left over after a failure will be processed. At startup, a recovery daemon detects temporary files that have been tagged for recovery.

The Media Content Recovery extension utilizes the `<data>` element to enable the application to associate recovery data with specific recorded content. The browser supports recovery data in the form of a completely specified HTTP URI. It is the VoiceXML application's responsibility to make sure the URI is correct and contains the information needed to deliver the content to the ultimate recipient.

If the Media Content Recovery extensions are not used, the IP Media Server processes the recording in accordance with standard VoiceXML. If the VoiceXML Media Content Recovery extensions are present in the VoiceXML script, then the recovery feature is used. A VoiceXML script that provides the user with a confirmation probably should not include the VoiceXML extensions due to the possibility that the recording could be recovered and sent without the user providing confirmation.

## VoiceXML Elements Reference

This section describes the VoiceXML elements (tags) supported by the Dialogic® IP Media Server. For a more detailed description of VoiceXML attributes, see [“VoiceXML Attributes Reference” \(page 348\)](#).

### <assign>

Assigns a value to a variable.

The value of this tag must be empty.

**Parents:** [<block>](#), [<catch>](#), [<error>](#), [<filled>](#), [<help>](#), [<if>](#), [<noinput>](#), [<nomatch>](#)

**Attributes:**

Attribute	Description
name	Name of the variable being assigned to a value. <b>Required.</b>
expr	ECMAScript expression that, when evaluated, is assigned as the new value of the variable. It must be a valid ECMA value. <b>Required.</b>

### <audio>

Plays pre-recorded audio and accesses the audio from the URI, which is specified by the src attribute.

The audio encoding is determined from the file header when .wav or .au formats are used. Otherwise, the audio encoding is implied by the file extension according to the following table.

Extension	Encoding
ulaw	G.711 $\mu$ -law
alaw	G.711 A law
msgsm	Microsoft® GSM

If the encoding cannot be determined by the extension, it is assumed to be G.711  $\mu$ -law.

The value of this tag can be empty or can contain one or more of the child elements listed below.

**Parents:** [<audio>](#), [<block>](#), [<catch>](#), [<choice>](#), [<error>](#), [<field>](#), [<filled>](#), [<help>](#), [<if>](#), [<initial>](#), [<menu>](#), [<noinput>](#), [<nomatch>](#), [<prompt>](#), [<record>](#), [<subdialog>](#)

**Child Elements:** [<audio>](#) [<break>](#) [<sayas>](#) [<value>](#)

**Attributes:**

Attribute	Description
src	String literal containing the URI for the audio. <b>Required.</b>

## <block>

Defines a container for non-interactive executable content such as welcome prompts.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parent:** [<form>](#)

**Child Elements:** [<assign>](#), [<audio>](#), [<clear>](#), [<disconnect>](#), [<exit>](#), [<goto>](#), [<if>](#), [<prompt>](#), [<reprompt>](#), [<return>](#), [<script>](#), [<submit>](#), [<throw>](#), [<value>](#), [<var>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
name	Name of the form-item variable used to track whether this block can be executed. Defaults to an inaccessible internal variable.

## <break>

Adds a pause to the audio content.

The value of this tag must be empty.

**Parents:** [<audio>](#), [<choice>](#), [<prompt>](#)

**Attributes:**

Attribute	Description
size	A relative pause duration, with possible values of none (0 ms), small (50 ms), medium (200 ms) or large (500 ms). The default is medium.
msecs	Text. The number of milliseconds to pause.

**<catch>**

Defines an event handler within the current scope.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parents:**     <field>, <form>, <initial>, <menu>, <record>, <subdialog>, <vxml>

**Child Elements:**   <audio> <value>

**Attributes:**

Attribute	Description
event	Event or events to catch. <b>Required.</b>
count	The occurrence of the event. (Default: 1.) The count allows you to handle different occurrences of the same event differently. Each form item and <menu> maintains a counter for each event that occurs while it is being visited; these counters are reset each time the <menu> or form item's <form> is re-entered.
cond	Boolean value of TRUE or FALSE. (Default: TRUE). When FALSE, the tag is ignored or skipped; the form is not visited.

**<choice>**

Defines a menu item by specifying one choice in the menu. It can specify a DTMF grammar fragment or the URI to go to when the choice is selected. When the choice is made, the tag either transitions to a new dialog or throws an event.

The Dialogic® IP Media Server handles DTMF grammar fragments only. The contents must be <audio> and not text.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parent:**       <menu>

**Child Elements:**   <audio>, <sayas>, <value>

**Attributes:**

Attribute	Description
dtmf	DTMF sequence for the choice.
event	An event to be thrown instead of going to ext.
expr	Expression to evaluate instead of going to next.
next	URI of next dialog or document.



To indicate the action to take, set one of the following: next, expr, or event.

## <clear>

Resets one or more form items to its initial state, including setting the form item variable to ECMAScript undefined and re-initializing the prompt counter and the event counters for the form item.

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attributes:**

Attribute	Description
namelist	List of items to clear. (When not specified, the application clears all form items in the current form.) <b>Required.</b>

## <data>

Enables [Media Content Recovery Extension](#). The Media Content Recovery feature is triggered by a <data> element with either of the following set:

- ◆ The src property set to builtin:persist.  
or
- ◆ The srcexpr property evaluates to the ECMA string builtin:persist.

An error.semantic event is generated if the <data> element is encountered with src or srcexpr set to anything but builtin:persist.

An error.semantic is generated if the <data> tag has any of the following attributes set with the builtin:persist data submission:

- ◆ method, enctype, fetchaudio, fetchint, fetchtimeout, maxage, maxstale

A valid <data> tag causes the recovery file to be created. If the recovery file cannot be written, an error.semantic event is generated.

The <data> tag is controlled entirely by its namelist attribute.

The namelist attribute must have one and only one variable name that is the same as the <record> element to be protected by the Media Content Recovery feature. The <record> input item must be in the same <form> element. If it is not, an error.semantic event is generated.

The namelist attribute must have one and only one recovery\_uri variable name that evaluates to a defined ECMA value that is not an array or an object. Without the recovery\_uri variable name the recovery file will not be written. The recovery daemon attempts content recovery by executing an HTTP POST method to the HTTP URI specified in the recovery information. When a file is being recovered, the recovery daemon uses the recording\_name variable as the name of the content in the HTTP POST.

The recovery file is deleted if a new <data> element is encountered that does not have the recovery\_uri name in the namelist. This allows an application to immediately disable the recovery feature if the user confirms that he does not want the message sent.

---

Note: Other uses of <data> as part of VoiceXML 2.1 are not supported.

---

**Parents:**     <block>, <if>, <filled>, <form>, <noinput>

The <data> tag must be a descendant of a <form> tag that contains the associated <record> tag.

**Child Elements:**   None.

**Attributes:**

Attribute	Description
namelist	See above.

---

Note: The Media Content Recovery extension currently supports audio content only.

---

### <disconnect>

Causes the interpreter context to disconnect from the user. As a result, the interpreter context generates an event (telephone.disconnected.hangup). Upon receiving the event, the application then performs a cleanup.

<disconnect> is the only supported VoiceXML call control tag (the same effect as <exit>).

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <noinput>, <nomatch>

**Attributes:**   None

### <dtmf>

Specifies a touch-tone grammar that defines both a set of key presses so a user can perform actions or supply information and the corresponding string values that describe those actions or information.

The value of this tag must be parsed text.

**Parents:**     <field>, <form>, <link>

**Attributes:**

Attribute	Description
src	URI for the source for an external grammar.

Attribute	Description
scope	Options are document, which makes the grammar active in all dialogs of the current document (and relevant application leaf documents), or dialog, which makes the grammar active throughout the current form. If omitted, the scope is determined by the parent element.
type	x-dtmf or regex

**<else>**

Marks the beginning of content to be executed when the parent `<if>` tag and all `<elseif>` tags at the same level of nesting have conditions that evaluate to false. Ends at the closing `<if>`.

The value of this tag must be empty.

**Parent:** `<if>`

**Attributes:** None.

**<elseif>**

Marks the beginning of content to be executed when the parent `<if>` and all `<elseif>` tags evaluate to false and the `cond` attribute evaluates to true. Ends at the next `<elseif/>` or `<else>` or closing `<if>` tag, whichever comes first.

The value of this tag must be empty.

**Parent:** `<if>`

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.

**<error>**

This element receives (catches) an error event. It is shorthand for `<catch event="error">` and receives all events of type 'error'.

The value can be empty or can contain one or more child elements listed below, where the element is either parsed text or the tag.

**Parents:** `<field>`, `<form>`, `<initial>`, `<menu>`, `<record>`, `<subdialog>`, `<vxml>`

**Child Elements:** `<assign>`, `<audio>`, `<clear>`, `<disconnect>`, `<exit>`, `<goto>`, `<if>`, `<prompt>`, `<reprompt>`, `<return>`, `<script>`, `<submit>`, `<throw>`, `<value>`, `<var>`

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
count	The occurrence of the event. (Default is 1.) The count allows you to handle different occurrences of the same event differently. Each form item and <menu> maintains a counter for each event that occurs while it is being visited; these counters are reset each time the <menu> or form item's <form> is re-entered.

**<exit>**

Halts all loaded documents and returns control to the interpreter context. Once <exit> returns control to the interpreter context, the interpreter context is free to do as it wishes. For example, it can play a top level menu for the user, drop the call, or the user to an operator.

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attributes:**   The IP Media Server returns no values and ignores the values specified in the expr or namelist attributes if they are supplied.

**<field>**

An input field that collects prompt-solicited user input within a form. Acceptable input is specified by type.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parent:**       <form>

**Child Elements:**   <audio>, <catch>, <dtmf>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>, <prompt>, <property>, <value>

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.

Attribute	Description
modal	If FALSE (the default), all active grammars are turned on while collecting this field. If TRUE, only the field's grammars are enabled: all others are temporarily disabled.
name	Field-item variable in the dialog scope that holds the result.
slot	Name of the grammar slot used to populate the variable. If it is absent, it defaults to the variable name.
type	ype of field, for example digits, currency, phone, date, Boolean, number, time. If not present, <grammar> and/or <dtmf> elements can be specified instead.

### <filled>

Specifies an action to perform when some combination of fields are filled by user input. It can occur in two places: as a child of the <form> element, or as a child of a field item.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parents:**     <field>, <form>, <record>, <subdialog>

**Child Elements:**   <assign>, <audio>, <clear>, <disconnect>, <exit>, <goto>, <if>, <prompt>, <reprompt>, <return>, <script>, <submit>, <throw>, <value>, <var>

**Attributes:**

---

Note: Attributes are valid only when <filled> is a child of <form>.

---

Attribute	Description
mode	Any or all. If any, action is executed when any field is filled by last user input, If all, action executes when all fields are filled.
namelist	The names of the form's field items.

### <form>

One of two kinds of dialogs for collecting user input. The other is <menu>.

The value of this tag can be empty or can contain one of more of the child tags listed below.

**Parent:**       <vxml>

**Child Elements:**   <block>, <catch>, <dtmf>, <error>, <field>, <filled>, <help>, <initial>, <link>, <noinput>, <nomatch>, <property>, <record>, <subdialog>, <var>

**Attributes:**

Attribute	Description
id	Name of the form.
scope	Default scope of the form's grammars.

**<goto>**

Used in executable content to cause a transition to another form item in the current form, another dialog in the current document, or another document.

The value of this tag must be empty.

**Parents:** [<block>](#), [<error>](#), [<filled>](#), [<help>](#), [<if>](#), [<noinput>](#), [<nomatch>](#)

**Attributes:** One of the following values must be specified:

Attribute	Description
expr	ECMAScript expression that yields the target URI.
expritem	ECMAScript expression that yields the name of the target form item.
next	URL to which to transition.
nextitem	Name of the next form item in the current form.
fetchtimeout	Interval to wait for the content to be returned before generating an error.badfetch event. If not specified, a value derived from the innermost fetchtimeout property is used.

**<grammar>**

This element is identical to [<dtmf>](#), because the Dialogic® IP Media Server supports only DTMF grammars.

**<help>**

Receives a help event, and is shorthand for [<catch event="help">](#). On the Dialogic® IP Media Server, there is currently no way for DTMF input to automatically throw a help event, unlike a speech recognition interface. The [<help>](#) tag is fully supported because an explicit grammar can define a DTMF sequence that causes the help event to be generated.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parents:** [<field>](#), [<form>](#), [<initial>](#), [<menu>](#), [<record>](#), [<subdialog>](#), [<vxml>](#)

**Child Elements:** [<assign>](#), [<audio>](#), [<clear>](#), [<disconnect>](#), [<else>](#), [<elseif>](#), [<exit>](#), [<goto>](#), [<if>](#), [<prompt>](#), [<reprompt>](#), [<return>](#), [<script>](#), [<submit>](#), [<throw>](#), [<value>](#), [<var>](#)

**Attributes:**

Attribute	Description
count	The occurrence of the event. (Default is 1.) The count allows you to handle different occurrences of the same event differently. Each form item and <menu> maintains a counter for each event that occurs while it is being visited; these counters are reset each time the <menu> or form item's <form> is re-entered.
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.

## <if>

Used for conditional logic. It has optional <else> and <elseif> elements.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parents:** [<block>](#), [<catch>](#), [<error>](#), [<filled>](#), [<help>](#), [<if>](#), [<noinput>](#), [<nomatch>](#)

**Child Elements:** [<assign>](#), [<audio>](#), [<clear>](#), [<disconnect>](#), [<else>](#), [<elseif>](#), [<exit>](#), [<goto>](#), [<if>](#), [<prompt>](#), [<reprompt>](#), [<return>](#), [<script>](#), [<submit>](#), [<throw>](#), [<value>](#), [<var>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.

## <initial>

Declares initial logic upon entry into a mixed-initiative form.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parent:** [<form>](#)

**Child Elements:** [<audio>](#), [<catch>](#), [<error>](#), [<help>](#), [<link>](#), [<noinput>](#), [<nomatch>](#), [<prompt>](#), [<property>](#), [<value>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
name	The name of a form-item variable used to track whether the <initial> is eligible to execute; defaults to an inaccessible internal variable.

**<link>**

Specifies a destination for a transition or an event to be thrown when input matches.

The value of this tag can be empty or can contain one or more grammar tags.

**Parents:**     [<dtmf>](#), [<field>](#), [<form>](#), [<initial>](#), [<vxml>](#)

**Attributes:** Must specify one of next, expr, or event.

Attribute	Description
next	URI to go to. Either a document (perhaps with an anchor to specify the starting dialog), or a dialog in the current document (just a bare anchor).
expr	Like next, except that the URI is dynamically determined by evaluating the given ECMAScript expression.
event	The event to generate when the user matches one of the link grammars.
fetchtimeout	Interval to wait for the content to be returned before generating an error.badfetch event. If not specified, a value derived from the innermost fetchtimeout property is used.

Dialogic's VoiceXML browser uses link grammars to support skip forward, skip back, and pause operations (VCR controls) for audio prompts.

**<log>**

When the log dest attribute is set to a value of SNMP, the VoiceXML browser sends out the msVXMLCriticalError trap to all SNMP trap hosts configured on the IP Media Server.



**Attributes:**

Attribute	Description
dest	<p>When set to SNMP, this instructs the browser to send out the msVXMLCriticalError trap to all SNMP trap hosts configured on the IP Media Server.</p> <p>The text value of the &lt;log&gt; element can be used to contain the details about the error. This content is placed in the msVXMLLastCriticalError object. This object is passed as a varbind of the msVXMLCriticalError trap.</p> <p>Following is an example &lt;log&gt; tag with the trap extension:</p> <pre>&lt;log dest="snmp"&gt;Unable to contact VoiceXML application server.&lt;/log&gt;</pre>

**<menu>**

Dialog for making a selection among choices.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parents:**     <vxml>

**Child Elements:**   <audio>, <catch>, <choice>, <error>, <help>, <noinput>, <nomatch>, <prompt>, <property>, <value>

**Attributes:**

Attributes	Description
id	Identifier of the menu. It allows the menu to be the target of a <goto> or a <submit>.
scope	Menu's grammar scope. Either dialog (the default) or document.
dtmf	TRUE or FALSE. When TRUE, any choices that do not have explicit DTMF elements are given implicit ones: 1, 2, etc.

**<meta>**

Page properties as meta-data, as in HTML.

---

**Note:** This is ignored by the VoiceXML Interpreter.

---

The value of this tag must be empty.

**Parent:**     <vxml>

**Attributes:**

Attribute	Description
name	Name of the meta-data property.
http-equiv	Name of an HTTP response header. Either name or http-equiv must be specified, but not both.
content	Value of the meta-data property. <b>Required.</b>

**<noinput>**

A type of catch element event used when the user does not respond within the required timeout interval. For example, <noinput> Shorthand for <catch event="noinput">. See [<catch>](#).

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parents:**     [<field>](#), [<form>](#), [<initial>](#), [<menu>](#), [<record>](#), [<subdialog>](#), [<vxml>](#)

**Child Elements:**   [<assign>](#), [<audio>](#), [<clear>](#), [<disconnect>](#), [<exit>](#), [<goto>](#), [<if>](#), [<prompt>](#), [<reprompt>](#), [<return>](#), [<script>](#), [<submit>](#), [<throw>](#), [<value>](#), [<var>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
count	A number that allows you to issue different prompts if the user is doing something repeatedly. If omitted, it defaults to 1.

**<nomatch>**

Used to catch a nomatch event. Shorthand for <catch event="nomatch">. See [“<catch>” \(page 328\)](#).

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parents:**     [<field>](#), [<form>](#), [<initial>](#), [<menu>](#), [<record>](#), [<subdialog>](#), [<vxml>](#)

**Child Elements:**   [<assign>](#), [<audio>](#), [<clear>](#), [<disconnect>](#), [<exit>](#), [<goto>](#), [<if>](#), [<prompt>](#), [<reprompt>](#), [<return>](#), [<script>](#), [<submit>](#), [<throw>](#), [<value>](#), [<var>](#)

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
count	A number that allows you to issue different prompts if the user is doing something repeatedly. If omitted, it defaults to 1.

**<param>**

Specifies the values that are passed to subdialogs.

The value of this tag must be empty.

**Parent:**     [<subdialog>](#)

**Attributes:**

Attribute	Description
name	Name to be associated with this parameter when the subdialog is invoked. <b>Required.</b>
expr	Expression that computes the value associated with name.
value	Associates a literal string value with name.
valuetype	Whether the value associated with name is data or a URI (ref).
type	MIME type of the result provided by a URI if the value type is ref; only relevant for uses of <param> in <object>.

**<prompt>**

Controls the output of prerecorded audio only.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parents:**     [<block>](#), [<catch>](#), [<error>](#), [<field>](#), [<filled>](#), [<help>](#), [<if>](#), [<initial>](#), [<menu>](#), [<noinput>](#), [<nomatch>](#), [<return>](#), [<subdialog>](#)

**Child Elements:**   [<audio>](#), [<break>](#), [<sayas>](#), [<value>](#)

**Attributes:**

Attribute	Description
bargein	Whether a prompt may be interrupted. (Default: TRUE.)
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.

Attribute	Description
count	A number that allows you to issue different prompts if the user is doing something repeatedly. If omitted, it defaults to 1.
timeout	How long to wait for the next input. The default noinput timeout is 5 seconds.
VCR	Whether VCR controls are to be active for the prompt block. (Default: FALSE.)

**<property>**

Sets a property value that affects platform behavior.

The value of this tag must be empty.

**Parents:** [<field>](#), [<form>](#), [<initial>](#), [<menu>](#), [<record>](#), [<subdialog>](#), [<vxml>](#)

**Attributes:**

Attribute	Description
name	Property name. <b>Required.</b>
value	Property value. <b>Required.</b>

**<record>**

A field item that collects a recording from the user. The recording is stored in the field item variable, which can be played back or submitted to the server. Only external storage is available.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parent:** [<form>](#)

**Child Elements:** [<audio>](#), [<catch>](#), [<error>](#), [<filled>](#), [<help>](#), [<noinput>](#), [<nomatch>](#), [<prompt>](#), [<property>](#), [<value>](#)

**Attributes:**

Attribute	Description
beep	TRUE or FALSE (the default). If TRUE, the application plays a tone just prior to recording.
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.

Attribute	Description
dtmfterm	TRUE or FALSE. If TRUE (default), a DTMF keypress terminates the recording.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
finalsilence	Interval of silence that indicates the end of speech. (Default: 3 seconds.)
max-time	Maximum duration to record. (Default: 10 seconds.)
name	The name of a form-item variable used to track whether the <initial> is eligible to execute; defaults to an inaccessible internal variable.
type	MIME type of the recording.

### <reprompt>

Used inside executable content to set a flag indicating that a new attempt should be made to issue a prompt for the current item.

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attributes:** None.

### <return>

Returns execution of a subdialog and returns control and data to the calling dialog.

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attributes:**

Attribute	Description
event	Returns, then sends this event.
namelist	Variable names to be returned to the calling dialog.

### <sayas>

Type of text construct contained within the element that defines the way a word or phrase is spoken, including text generated by the <value> tag. The rules to convert <sayas> text into pre-recorded speech fragments are determined by

an external script. The default URI of this script is `http://localhost/snowshore/vxmlphrase.cgi`. This URI can be modified to use an external server by the `com.snowshore.recsrc` property.

The default rules handle US English only. The value of this tag must be parsed text.

**Parents:** `<audio>`, `<choice>`, `<prompt>`

**Attributes:**

Attribute	Description
class	Describes the way to render the value of: <ul style="list-style-type: none"> <li>• phone (NANP numbers)</li> <li>• digits (Each spoken with no pauses.)</li> <li>• number (Spoken as a positive cardinal number. Negative numbers are not supported and will produce odd output.)</li> </ul>

## **<script>**

Allows the specification of a block of client-side ECMAScript code.



Note: Although the Dialogic® IP Media Server supports this tag, Dialogic does not recommend its use and considers it inadvisable to execute arbitrary logic on the IP Media Server due to resource limitations.

The value of this tag must be parsed text.

**Parents:** `<block>`, `<catch>`, `<error>`, `<filled>`, `<help>`, `<if>`, `<noinput>`, `<nomatch>`, `<vxml>`

**Attributes:**

Attribute	Description
src	URL for the resource.

## **<subdialog>**

Invokes a second dialog within the current dialog. It provides a way for invoking a new interaction, and returning to the original form. Local data, grammars, and state information are saved and available when returning to the calling document.

The value can be empty or can contain one or more of the child elements listed below, where the element is either parsed text or the tag.

**Parent:** `<form>`

**Child Elements:** `<audio>`, `<catch>`, `<error>`, `<filled>`, `<help>`, `<if>`, `<noinput>`, `<nomatch>`, `<param>`, `<prompt>`, `<property>`, `<value>`

**Attributes:**

Attribute	Description
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
method	Request method (get or post).
modal	Controls which grammars are active during the subdialog.
name	The result returned from the subdialog, an ECMAScript object whose properties are the ones defined in the namelist attribute of the <return> element.
namelist	List of variables to submit to the subdialog. Same as namelist in <submit>, except that the default is to submit nothing. Only valid when fetching another document.
src	URI of the subdialog. <b>Required.</b>

**<submit>**

This element is like <goto> because the application gets a new document. Unlike <goto>, it lets you submit a list of variables to the document server, using an HTTP GET or POST request.

The value of this tag must be empty.

**Parents:**    <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attributes:**

Attribute	Description
expr	Like next, except that the URI is dynamically determined by evaluating the given ECMAScript expression. One of next, expr, or fetchtimeout is required.
method	The request method (GET or POST). The default is GET.
namelist	The list of variables to submit.
next	The URL to which the query is submitted.
fetchtimeout	Interval to wait for the content to be returned before generating an error.badfetch event. If not specified, a value derived from the innermost fetchtimeout property is used.

**<throw>**

Sends an event to be received by <catch>.

The value of this tag must be empty.

**Parents:**     <block>, <catch>, <error>, <filled>, <help>, <if>, <noinput>, <nomatch>

**Attribute:**

Attribute	Description
event	Returns, then sends this event.

**<transfer>**

Provides the ability to place an outgoing voice call to support transfers. A “bridged” transfer session returns the caller to the original session with the interpreter upon completion. A “blind” transfer session terminates when completed.

**Parent:**       <form>



**Attributes:**

Attribute	Description
name	The outcome of the transfer attempt. For possible return values, see the following table.
expr	Initial value of the form-item variable; the default is ECMAScript undefined. If initialized to a value, then the form item is not visited unless the form-item variable is cleared.
cond	Boolean value of TRUE or FALSE. Defaults to TRUE. When FALSE, the tag is ignored or skipped; the form is not visited.
dest	The URL of the destination (phone, IP telephone address).
destexpr	An ECMAScript expression yielding the URI of the destination.
bridge	What to do once the call is connected. If bridge is TRUE, document interpretation suspends until the transferred call terminates. Blind transfer is supported by hairpinning RTP through the IP Media Server.
connect	Time to wait while trying to connect the timeout call before returning the noanswer condition. Default is platform specific.
maxtime	Time that the call is allowed to last, or 0 if it can last arbitrarily long. Only applies if bridge is TRUE. (Default: 0.)
transfer	URI of audio source to play while the audio transfer attempt is in process (before far-end answer). If the resource cannot be fetched, the error is ignored and the transfer continues.
reqUri	SIP URI to be used as the Request-URI in the outbound INVITE. The value must be compliant with the grammar for a SIP URL. If this attribute is not present the Request-URI will have the same value as the To header. The To URI value is determined by the dest attribute.
requi expr	An ECMAScript expression that evaluates to a SIP URI.
src	The SIP URI that shows up as ANI at the callee. Must be of the form sip:user@server. This is to identify the caller for billing purposes. (Dialogic-specific attribute)
srcexpr	An ECMAScript expression yielding src. (Dialogic-specific attribute).
stopdigits	Defines the list of single DTMF digits that ends the transfer.
longdigit	Enables detection of long DTMF digits or multiple, short instance of the same digit.
video	Whether video is available on the outbound call. The value can be one of suppress or allow. See <a href="#">“RTP Codec Selection with VoiceXML &lt;transfer/&gt;”</a> (page 378).

Note: The src, srcexpr, stopdigits, longdigit, and video attributes are Dialogic-specific attributes.

Return values of Name variable:

Value	Description
busy	The endpoint refused the call.
noanswer	There was no answer within the specified time.
network busy	Some intermediate network refused the call.
near_end_disconnect	The call completed and was terminated by the caller.
far_end_disconnect	The call completed and was terminated by the callee.
network_disconnect	The call completed and was terminated by the network.

### <value>

Inserts the value of an ECMAScript expression into a prompt.

The value of this tag must be empty.

**Parents:** [<audio>](#), [<block>](#), [<catch>](#), [<choice>](#), [<error>](#), [<field>](#), [<filled>](#), [<help>](#), [<if>](#), [<initial>](#), [<menu>](#), [<noinput>](#), [<nomatch>](#), [<record>](#), [<subdialog>](#)

**Attributes:**

Attribute	Description
expr	Expression to render. <b>Required.</b>
class	Data type of the value.
recsrc	URI used to determine text to prompt conversion. Defaults to the value of com.snowshore.recsrc if specified, otherwise it is "http://localhost/snowshore/phrase.cgi?".

### <var>

Declares an ECMAScript variable. The scope of the variable is determined by the parent tag.

The value of this tag must be empty.

**Parents:** [<block>](#), [<catch>](#), [<error>](#), [<filled>](#), [<form>](#), [<help>](#), [<if>](#), [<noinput>](#), [<nomatch>](#)

**Attributes:**

Attribute	Description
name	Name of the variable. <b>Required.</b>
expr	An initial value.

**<vxml>**

The top-level element in a VoiceXML document.

The value of this tag must contain one or more of the child elements listed below.

**Parent:** None

**Child Elements:** [<catch>](#), [<error>](#) [<form>](#), [<help>](#), [<link>](#), [<menu>](#), [<meta>](#), [<noinput>](#), [<nomatch>](#), [<property>](#), [<script>](#), [<var>](#)

**Attributes:**

Attribute	Description
version	Version of VoiceXML. <b>Required.</b>
base	Base URL. The root document URL, if any.

## VoiceXML Attributes Reference

This section describes the VoiceXML attributes supported by the Dialogic® IP Media Server.

### application

Specifies the URI of the root document.

**Attribute for:** `<vxml>`

### bargein

Specifies whether user input can interrupt a prompt.

The IP Media Server supports the energy bargein type. The prompt is stopped if a DTMF tone is detected.

**Attribute for:** `<prompt>`

**Values:**

Value	Description
true (default)	Allow input to interrupt.
false	Do not allow input to interrupt.

### bargeintype

Type of interrupt (bargein) the IP Media Server performs in response to DTMF inputs.

**Attribute for:** `<prompt>`

**Values:**

Value	Description
true	The default value of bargein is true.
false	The default value of bargein is false.

---

Note: Dialogic recommends running several tests using all combinations of the prompt element and its attributes.

---

### base

The base URL for interpreting relative URLs.

**Attribute for:** `<vxml>`

## beep

Whether to play a tone before recording.

**Attribute for:** `<record>`

**Values:**

Value	Description
true	Plays a tone when recording starts.
false (default)	Does not play tone when recording starts.

## bridge

Controls the platform behavior once the call is connected. If set to `true`, the application stops interrupting documents until the transferred call terminates, at which time document processing resumes.

If set to `false`, document interpretation is also suspended but terminates immediately after the transferred call ends. This behavior does not follow the VoiceXML specification which calls for the interpreter to throw a `telephone.disconnect.transfer` event as soon as the call connects. Bridged transfer is currently implemented by hair-pinning media from both calls through the IP Media Server, which requires the VoiceXML session to continue its existence for the lifetime of the call transfer.

**Attribute for:** `<transfer>`

**Values:**

Value	Description
true	Suspends document interpretation until the transferred call terminates, at which time document processing resumes.
false (default)	Suspends document interpretation but terminates immediately after the transferred call ends.

## class

The data type of the value.

**Attribute for:** `<sayas>` `<value>`

**Values:**

Value	Description
phone	The value is spoken as a phone number, with a leaning toward the North American Numbering Plan (NANP). 10 digit numbers have breaks after the 3rd and 6th digits. 7 digit numbers have breaks after the 3rd digit. 4 digit numbers are spoken as is. 11 digit numbers, where the first digit is a one, are spoken with a pause after the 1st, 4th, and 7th digit. Other length numbers are spoken with a pause after the 2nd digit, and each 3rd digit thereafter.
digits	Each digit is spoken with no pauses.
number	The value is spoken as a positive cardinal number. Negative numbers are not supported and will produce odd output.

**cond**

Boolean ECMAScript expression that serves as a guard.

**Attribute for:** [<block>](#), [<catch>](#), [<elseif>](#), [<error>](#), [<field>](#), [<help>](#), [<if>](#), [<initial>](#), [<noinput>](#), [<nomatch>](#), [<prompt>](#), [<record>](#), [<subdialog>](#), [<transfer>](#)

**Values:**

Value	Description
true (default)	The form item is executed.
false	The form item is ignored or skipped; the form is not visited.

**connect-timeout**

Time to wait while trying to connect the call before returning a noanswer condition.

**Attribute for:** [<transfer>](#)

**Values:** Positive integers expressed in milliseconds or seconds when a qualifier “s” is appended to the value, for example 1s.

**content**

Value of the meta-data property.

**Attribute for:** [<meta>](#)

**count**

Repeat count for prompt or handler.

**Attribute for:** [<catch>](#), [<error>](#), [<help>](#), [<noinput>](#), [<nomatch>](#), [<prompt>](#)

**Values:**

- ◆ For prompts, the minimum count for the prompt to be played. Set to zero when dialog is initialized. Increases by one each time user is prompted. The `<clear>` tag resets it to zero.
- ◆ For event handlers, the minimum count for the handler to be eligible to handle an event. Set to zero at initialization and increased by one each time event is triggered.

## dest

Destination address for outbound calls initiated using `<transfer/>`.

**Attribute for:** [<transfer>](#)

**Value:** The `dest` attribute must be in the form of a valid SIP URI, for example `sip:19783678400@gateway.carrier.net`. (Default: none.)

## destexpr

An ECMAScript expression yielding the destination address for outbound calls initiated via `<transfer/>`.

**Attribute for:** [<transfer>](#)

**Value:** The expression must produce a valid SIP URI, for example `sip:19783678400@gateway.carrier.net`. (Default: none.)

## dtmf

Specifies touch-tone.

**Attribute for:** [<choice>](#), [<menu>](#)

- ◆ `choice` Text /CDATA (touch-tone digit for this choice).
- ◆ `menu` Dialog for making a selection among choices. See “[<menu>](#)” (page 337).

**Values:**

Value	Description
true	Assigns touch-tone digits.
false	Does not assign touch-tone digits.

## dtmfterm

Whether to allow touch-tone interruption.

**Attribute for:** [<record>](#)

**Values:**

Value	Description
true (default)	Touch-tone terminates recording.
false	Touch-tone does not terminate recording.

**event**

Name of an event.

**Attribute for:** [<catch>](#), [<choice>](#), [<link>](#), [<return>](#), [<throw>](#)

**expr**

ECMAScript expression to evaluate. Depending on the tag, used to set a variable, a URL, or for other purposes.

**Attribute for:** [<assign>](#), [<audio>](#), [<block>](#), [<choice>](#), [<exit>](#), [<field>](#), [<goto>](#), [<initial>](#), [<link>](#), [<param>](#), [<record>](#), [<subdialog>](#), [<submit>](#), [<value>](#), [<var>](#), [<transfer>](#)

**Value:** Text (CDATA).

**exritem**

ECMAScript that provides item name.

**Attribute for:** [<goto>](#)

**Value:** Text (CDATA).

**finalsilence**

Interval of silence that indicates end of speech.

**Attribute for:** [<record>](#)

**Value:** Text (CDATA). (Default: 3 seconds.)

**http-equiv**

The HTTP response header field name.

**Attribute for:** [<meta>](#)

**Value:** Text (NMTOKEN)

**id**

Names an element for later reference.



**Attribute for:** [<form>](#), [<menu>](#)

**Value:** URL#anchortname is the syntax.

## longdigit

Turns detection on or off on long DTMF digits or multiple, short instances of the same digit in a two-second interval.

**Attribute for:** [<transfer>](#)

**Values:**

Value	Description
yes	Turns DTMF detection on for long digits.
no (default)	Turns DTMF detection off for long digits.

## max-time

Maximum time the call is allowed to last, or 0 (zero) if call duration is unlimited. This attribute only applies when the `bridge` attribute is set to `true`.

**Attribute for:** [<transfer>](#)

**Values:** Positive integers expressed in milliseconds or seconds (when a qualifier “s” is appended to the value, for example 1s). (Default: 0. (zero))

## method

The request method.

**Attribute for:** [<subdialog>](#) [<submit>](#)

**Values:**

Value	Description
get (default)	HTTP GET method.
post	HTTP POST method.

## modal

Whether to enable outside grammars.

**Attribute for:** [<field>](#), [<subdialog>](#)

**Values:**

Value	Description
true (default)	Disable higher-level grammars in the scope.
false	Enable higher-level grammars in the scope.

**mode**

Specifies mode for performing action when fields are filled in a form.

**Attribute for:** [<filled>](#)

**Values:**

Value	Description
all (default)	Action executes when all fields are filled.
any	Action is executed when any field is filled by user input.

**msecs**

Number of milliseconds to pause.

**Attribute for:** [<break>](#)

**Value:** Text. A length of time in milliseconds as a numerical value.

**name**

Name of an item, variable, or parameter.

**Attribute for:** [<assign>](#), [<block>](#), [<field>](#), [<initial>](#), [<meta>](#), [<param>](#), [<property>](#), [<record>](#), [<subdialog>](#), [<var>](#), [<transfer>](#)

**Values:** Text.

**namelist**

List of variable names.

**Attribute for:** [<clear>](#), [<data>](#), [<exit>](#), [<filled>](#), [<return>](#), [<subdialog>](#), [<submit>](#)

**Value:** A white space-separated list of variable or field names.

## next

URL of the next page or dialog. This element is similar to `<goto>`, because it results in a new dialog being obtained. Unlike `<goto>`, it lets you submit a list of variables to the document server using an HTTP GET or POST request.

**Attribute for:** `<choice>`, `<goto>`, `<link>`

## nextitem

First item to visit in the next dialog.

**Attribute for:** `<goto>`

**Value:** Text (NMTOKEN)

## recsrc

Method to use when concatenating audio.

**Attribute for:** `<value>`

**Value:** The value of `expr` is expanded into a list of audio files by a CGI script external to the VoiceXML process. The default URI of this script is `http://localhost/Cantata/vxmlphrase.cgi`. This URI can be modified to use an external server by the `com.Cantata.recsrc` property, or by setting the `recsrc` attribute. The class and value of the phrase to be rendered are sent as properties of the CGI script “class” and “value”.

## reqUri

SIP URI to be used as the Request-URI in the outbound INVITE. The value must be compliant with the grammar for a SIP URL. If this attribute is not present the Request-URI will have the same value as the To header. The To URI value is determined by the `dest` attribute.

## scope

Range where an element can be used.

**Attribute for:** `<form>`, `<grammar>`, `<menu>`

**Values:**

Value	Description
dialog	Scope is the current dialog.
document	Scope is the current document.

## size

Relative pause duration.

**Attribute for:** [<break>](#)

**Values:**

Value	Description
none	0 ms
small	50 ms
medium (default)	200 ms
large	500 ms

## slot

Name of the grammar slot used to populate the variable. If the grammar slot is absent, it defaults to the variable name. This attribute is useful when the grammar format being used has a mechanism for returning sets of slot/value pairs and the slot names differ from the field item variable names. If the grammar returns only one slot, as do the built-in type grammars like Boolean, then no matter what the slot's name, the field item variable gets the value of that slot.

**Attribute for:** [<field>](#)

**Value:** Text (NMTOKEN)

## src

Calling party identification (ANI) used when placing the outbound call through [<transfer>](#). This identifies the caller for billing purposes.

**Attribute for:** [<transfer>](#)

**Value:** Must be in the form of a valid SIP URI, for example subscriber103@vm.carrier.net. (Default: none.)

## srcexpr

ECMAScript expression yielding the calling party identification.

**Attribute for:** [<transfer>](#)

**Value:** The expression must produce a valid SIP URI, for example sip:subscriber103@vm.carrier.net.

## stopdigits

List of DTMF digits that terminate a transfer.

**Attribute for:** [<transfer>](#)

**Values:** 0-9, #, A-D. (Default: #.)

## timeout

Maximum time to wait before triggering a `<noinput>` event.

**Attribute for:** `<prompt>`

**Value:** Text. A numerical value with optional decimals, followed by "s" for seconds or "ms" for milliseconds.

## transfer-audio

URI of the audio source to play while the transfer attempt is in process. If the resource cannot be fetched, the application ignores the error and continues the transfer.

**Attribute for:** `<transfer>`

**Value:** Valid HTTP or file scheme URI. (No default.)

## type

Indicates how to interpret text to be spoken. The meaning of this attribute depends on the tag: for `dtmf`, `grammar`, `object`, `param`, and `record`, it is a content type (MIME type); for `field` and `say-as`, it is a data type.

**Attribute for:** `<dtmf>`, `<field>`, `<grammar>`, `<object>`, `<param>`, `<record>`, `<sayas>`

**Values:**

Value	Description
dtmf	x-dtmf or regex.
field	A built-in type (Boolean, digits, phone, date, currency, number, time).
grammar	Content type of the grammar.
object	Content type of the object data.
param	MIME type of the result provided by a URI if the value type is ref; only relevant for uses of <code>&lt;param&gt;</code> in <code>&lt;object&gt;</code> .
record	MIME type of the recording. See <a href="#">“MIME Recording Encoding Types” (page 359)</a> .
sayas	Data type to use in interpreting the value for the following: telephone, date, digits, number.

## value

Value to assign to the parameter or property named by the name attribute.

**Attribute for:**     <param>, <property>

## valuetype

Value to assign to the parameter or property named by the name attribute.

**Attribute for:**     <param>

**Values:**

Value	Description
data (default)	Value is data.
ref	Value is a URL pointing to data.

## version

Version of VoiceXML. Required.

**Attribute for:**     <vxml>

**Value:**            1.0.

## VCR

Whether VCR controls are active

**Attribute for:**     <prompt>

**Values:**

Value	Description
true	VCR controls are enabled.

## video

Whether video media is offered on outgoing calls.

**Attribute for:**     <transfer>

**Values:**

Value	Description
allow	Video media is offered on outgoing calls.
suppress	Video media is not offered on outgoing calls.

See “[RTP Codec Selection with VoiceXML <transfer/>](#)” (page 378) for additional information on the detailed interaction of the video attribute and RTP codec selection.

# VoiceXML Properties

## com.snowshore.criticaldigit\_timer

Sets a special digit timer that specifies the interval to wait for additional user input before returning a match to an active grammar. This property is used when the application modifies the DTMF matching behavior.

Uses the VoiceXML standard "n {ms} | {s}" timer notation. Default unit is milliseconds (ms).

### Values:

Value	Description
-2	The default critical digit timer is set to the same value as the interdigit timer (thus it has no effect).
-1	Immediate. Use "shortest match first".
0	Infinite. Use "longest match first", but waits infinitely once a match is found. Thus, matched input must end with a return digit or restart digit.
>0	Uses "longest match first", and waits that amount of time when a match is found.

## MIME Recording Encoding Types

The MIME type and optional codec parameters define the file format and audio encoding for the recording. Supports both headerless (raw) and .wav file formats. Table 58 lists the MIME type, file format, and audio encoding.

Table 58. Supported MIME Types, Formats, and Audio Encodings

MIME Type	File Format	Audio Encoding
audio / basic	raw	G.711 $\mu$ -law
audio/ x-alaw-basic	raw	G.711 A law

There are various commonly-used methods of specifying .wav format and the desired audio encodings. The official IANA registered MIME type for wave is audio/vnd.wave but audio/wav and audio/x-wav are used frequently. Table 59 lists the accepted WAVE format specifiers.

Table 59. Accepted WAVE Format Specifiers

MIME Type	File Format	Notes
audio/vnd.wave	WAVE	IANA standard
audio/wav	WAVE	
audio/x-wav	WAVE	Proprietary
video/x-wav	WAVE	

Specify audio encoding by adding a codec parameter and value to the base MIME type, for example:

```
audio/vnd.wave; codec=7
```

The official values for the codec parameter are defined by the IANA registry (<http://www.iana.org/assignments/wave-avi-codec-registry>), but other commonly used forms are also supported.

Table 60 provides a summary of the supported codec parameters and values and their corresponding audio encodings.

Table 60. Codec Parameter Values

Value	Audio Encoding	Notes
7	G.711 $\mu$ -law	IANA standard
6	G.711 A law	IANA standard
31	Microsoft® GSM	IANA standard
ulaw	G.711 $\mu$ -law	
alaw	G.711 A law	
msgsm	Microsoft® GSM	

The video/x-wav MIME type always produces a file with a G.711 audio track and an H.263 video track. No codec parameters are supported for this type.



# ECMAScript Functionality

Table 61 summarizes the categories and features supported by the IP Media Server's VoiceXML interpreter.

**Note:** The ECMAScript Language Specification can be downloaded for free.

Table 61. ECMAScript Functionality

Category	Feature/Keyword
Array Handling	Array join, length, reverse, sort
Assignments	Assign (=) Compound Assign (OP=)
Booleans	Boolean
Comments	/*...*/ or //
Constants/Literals	NaN null true, false Infinity undefined
Control flow	Break continue for for...in if...else return while
Dates and Time	Date getDate, getDay, getFullYear, getHours, getMilliseconds, getMinutes, getMonth, getSeconds, getTime, getTimezoneOffset, getYear, getUTCDate, getUTCDay, getUTCFullYear, getUTCHours, getUTCMilliseconds, getUTCMinutes, getUTCMonth, getUTCSeconds, setDate, setFullYear, setHours, setMilliseconds, setMinutes, setMonth, setSeconds, setTime, setYear, setUTCDate, setUTCFullYear, setUTCHours, setUTCMilliseconds, setUTCMinutes, setUTCMonth, setUTCSeconds, toGMTString, toLocaleString, toUTCString, parse, UTC

Table 61. ECMAScript Functionality (continued)

Category	Feature/Keyword
Declarations	Function new this var with
Function Creation	Function arguments, length
Global Methods	Global escape, unescape eval isFinite, isNaN parseInt, parseFloat
Math	Math abs, acos, asin, atan, atan2, ceil, cos, exp, floor, log, max, min, pow, random, round, sin, sqrt, tan, E, LN2, LN10, LOG2E, LOG10E, PI, SQRT1_2, SQRT2
Numbers	Number MAX_VALUE, MIN_VALUE NaN NEGATIVE_INFINITY, POSITIVE_INFINITY
Object Creation	Object new constructor, prototype, instanceof, toString, valueOf
Operators	Addition (+), Subtraction (-) Modulus arithmetic (%) Multiplication (*), Division (/) Negation (-) Equality (==), Inequality (!=) Less Than (<), Less Than or Equal To (<=) Greater Than (>) Greater Than or Equal To (>=) Logical And(&&), Or (  ), Not (!) Bitwise And (&), Or ( ), Not (~), Xor (^) Bitwise Left Shift (<<), Shift Right (>>) Unsigned Shift Right (>>>) Conditional (?:) Comma (,) delete, typeof, void Decrement (--), Increment (++)

Table 61. ECMAScript Functionality (continued)

Category	Feature/Keyword
Objects	Array Boolean Date Function Global Math Number Object String
Strings	String charAt, charCodeAt, fromCharCode indexOf, lastIndexOf split toLowerCase, toUpperCase length

## Support for VoiceXML Extended Session Variables

The Dialogic® VoiceXML browser supports the following VoiceXML 1.0 session variables:

- ◆ session.telephone.ani
- ◆ session.telephone.dnis
- ◆ session.telephone.rdnis (always set to NULL)
- ◆ session.telephone.redirect\_reason

In addition to the above, other session variables are supported so that a VoiceXML script can access a complete set of information from the SIP call. Some of the new session variables are defined in VoiceXML 2.0, and others are Dialogic® extensions as noted on the following page.

### VoiceXML 2.0 Recommendations

session.connection.local.uri

Value: Set to the SIP Request-URI; note that any SIP escapes (%xx) in the Request-URI are still in this variable as well.

session.connection.remote.uri

Value: Set to the SIP From-URI.

session.connection.protocol.name

Value: Always SIP.

session.connection.protocol.version

Value: Always 2.0.

### Dialogic® Extensions

session.connection.media

An array where each element represents an RTP media stream. Each array element has the following associated attributes:

- ◆ "type" - indicates the MIME type of media stream (currently either "audio" or "video")
- ◆ "subtype" - indicates the MIME sub-type, or encoding, of the media stream (e.g. PCMU, PCMA, H264, etc.)

session.connection.protocol.sip.parameter

Value: An array of name/value pairs parsed from the SIP Request-URI; the name and value are "UnEscaped," meaning any SIP escapes (%xx) in the URI are expanded before being assigned.

session.connection.protocol.sip.parameter[n].name

Value: The name of the nth parameter.

`session.connection.protocol.sip.parameter[n].value`

Value: The "value" (if any, can be empty) of the nth parameter.

`session.connection.protocol.sip.uri`

Value: Set to the SIP Request-URI. Note that any SIP escapes (%xx) in the Request-URI are still in this variable as well. This variable duplicates the `session.connection.local.uri` by design.

`session.connection.protocol.sip.to`

Value: Set to the SIP To-URI. Since the VoiceXML 2.0 defined `session.connection.local.uri` variable is mapped to the request URI a separate variable is required to contain the contents of the SIP To header.

`session.connection.protocol.sip.from`

Value: Set to the SIP From-URI. This variable duplicates the `session.connection.remote.uri` by design.

`session.connection.protocol.sip.call_id`

Value: Set to the SIP Call ID.

## Example

The following example shows the mapping between SIP headers and the newly supported VoiceXML session variables.

Given the following SIP INVITE request:

```
INVITE
sip:dialog@10.102.4.26;voicexml=http://10.102.4.9:9020
  /ivr/sip_init.vxml%3fdnis%3d961234567%26hasvideo%3d
  1;dogs=nice;user=phone
SIP/2.0
From: <sip:968037431@10.102.4.45:5060>;tag=2d04660a-
  13c4-40ed433f-9ec19fd-1786
To: <sip:961234567@10.102.4.9>
Call-ID: 6a7b774-2d04660a-13c4-40ed433f-9ec19fd-
  7061@10.102.4.45
CSeq: 1 INVITE
Via: SIP/2.0/UDP
  10.102.4.134:5060;branch=z9hG4bKi8MKi8i!yi8MK2SsMu8
  Uyake2q0OUqMi8i!y.1-1d9515c
Via: SIP/2.0/UDP
  10.102.4.45:5060;received=10.102.4.45;branch=z9hG4b
  K-40ed433f-9ec19fd-847
Max-Forwards: 69
User-Agent: RADVision ViaIP GW Vers. 1.0
Call-Info: <Media:Video>;purpose=info
Contact: <sip:968037431@10.102.4.45:5060>
Content-Length:0
```

The Media Server populates the new VoiceXML session variables, as shown:

```
session.connection.protocol.name = "SIP"
session.connection.protocol.version = "2.0"
session.connection.local.uri =
  sip:dialog@10.102.4.26;voicexml=http://10.102.4.9:9
  020/ivr/sip_init.vxml%3fdnis%3d961234567%26hasvideo
  %3dl;dogs=nice;user=phone"
session.connection.remote.uri =
  <sip:968037431@10.102.4.45:5060>;tag=2d04660a-13c4-
  40ed433f-9ec19fd-1786
session.connection.protocol.sip.parameter[0].name =
  "voicexml"
session.connection.protocol.sip.parameter[0].value =
  "http://10.102.4.9:9020/ivr/sip_init.vxml?dnis=9612
  34567&hasvideo=1"
session.connection.protocol.sip.parameter[1].name =
  "dogs"
session.connection.protocol.sip.parameter[1].value =
  "nice"
session.connection.protocol.sip.parameter[2].name =
  "user"
session.connection.protocol.sip.parameter[2].value =
  "phone"
session.connection.protocol.sip.uri =
  sip:dialog@10.102.4.26;voicexml=http://10.102.4.9:9
  020/ivr/sip_init.vxml%3fdnis%3d961234567%26hasvideo
  %3dl;dogs=nice;user=phone"
session.connection.protocol.sip.to =
  <sip:961234567@10.102.4.9>
session.connection.protocol.sip.from =
  <sip:968037431@10.102.4.45:5060>;tag=2d04660a-13c4-
  40ed433f-9ec19fd-1786
session.connection.protocol.sip.call_id = 6a7b774-
  2d04660a-13c4-40ed433f-9ec19fd-7061@10.102.4.45
```

## C - MSCML Schema

### Example 48. MSCML Schema

```
<?xml version="1.0" encoding="UTF-8"?>
<xs:schema xmlns:xs="http://www.w3.org/2001/XMLSchema"
  elementFormDefault="qualified">
  <xs:element name="MediaServerControl">
    <xs:complexType>
      <xs:choice>
        <xs:element name="request">
          <xs:complexType>
            <xs:choice>
              <xs:element name="configure_conference"
                type="configure_conferenceRequestType"/>
              <xs:element name="configure_leg"
                type="configure_legRequestType"/>
              <xs:element name="play" type="playRequestType"/>
              <xs:element name="playcollect"
                type="playcollectRequestType"/>
              <xs:element name="playrecord"
                type="playrecordRequestType"/>
              <xs:element name="managecontent"
                type="managecontentRequestType"/>
              <xs:element name="faxplay"
                type="faxRequestType"/>
              <xs:element name="faxrecord"
                type="faxRequestType"/>
              <xs:element name="stop" type="stopRequestType"/>
            </xs:choice>
          </xs:complexType>
        </xs:element>
        <xs:element name="response" type="responseType"/>
      </xs:choice>
    </xs:complexType>
  </xs:element>
</xs:schema>
```

```

    <xs:element name="notification">
      <xs:complexType>
        <xs:choice>
          <xs:element name="conference"
            type="conferenceNotificationType"/>
          <xs:element name="keypress"
            type="keypressNotificationType"/>
          <xs:element name="signal"
            type="signalNotificationType"/>
        </xs:choice>
      </xs:complexType>
    </xs:element>
  </xs:choice>
  <xs:attribute name="version" use="required"/>
</xs:complexType>
</xs:element>

<!-- Definitions for base and concrete MSCML requests -->
<!-- and embedded types. -->
<xs:complexType name="base_requestType" abstract="true">
  <xs:attribute name="id" type="xs:string"/>
</xs:complexType>
<xs:complexType name="playRequestType">
  <xs:complexContent>
    <xs:extension base="base_requestType">
      <xs:sequence>
        <xs:element name="prompt" type="promptType"
          minOccurs="0"/>
      </xs:sequence>
      <xs:attribute name="prompturl" type="xs:string"/>
      <xs:attribute name="offset" type="xs:string"/>
      <xs:attribute name="promptencoding" type="xs:string"/>
    </xs:extension>
  </xs:complexContent>
</xs:complexType>
<xs:complexType name="configure_conferenceRequestType">
  <xs:complexContent>
    <xs:extension base="base_requestType">
      <xs:sequence>
        <xs:element name="subscribe"
          type="conference_eventsubscriptionType" minOccurs="0"/>
      </xs:sequence>
      <xs:attribute name="reservedtalkers"
        type="xs:positiveInteger"/>
      <xs:attribute name="reserveconfmedia" type="yesnoType"
        default="yes"/>
    </xs:extension>
  </xs:complexContent>
</xs:complexType>
<xs:complexType name="configure_legRequestType">
  <xs:complexContent>
    <xs:extension base="base_requestType">

```



```
<xs:sequence>
  <xs:element name="inputgain" type="gainType"
    minOccurs="0" />
  <xs:element name="outputgain" type="gainType"
    minOccurs="0" />
  <xs:element name="configure_team"
    type="configure_teamType" minOccurs="0" />
  <xs:element name="subscribe"
    type="leg_eventsubscriptionType" minOccurs="0" />
</xs:sequence>
<xs:attribute name="type">
  <xs:simpleType>
    <xs:restriction base="xs:NMTOKEN">
      <xs:enumeration value="talker" />
      <xs:enumeration value="listener" />
    </xs:restriction>
  </xs:simpleType>
</xs:attribute>
<xs:attribute name="mixmode">
  <xs:simpleType>
    <xs:restriction base="xs:NMTOKEN">
      <xs:enumeration value="full" />
      <xs:enumeration value="mute" />
      <xs:enumeration value="preferred" />
      <xs:enumeration value="parked" />
      <xs:enumeration value="private" />
    </xs:restriction>
  </xs:simpleType>
</xs:attribute>
<xs:attribute name="dtmfclamp" type="yesnoType" />
<xs:attribute name="toneclamp" type="yesnoType" />
</xs:extension>
</xs:complexContent>
</xs:complexType>
<xs:complexType name="configure_teamType">
  <xs:sequence>
    <xs:element name="teammate" type="teammateType" minOccurs="0"
      maxOccurs="unbounded" />
  </xs:sequence>
  <xs:attribute name="id" type="xs:string" />
  <xs:attribute name="action" use="required">
    <xs:simpleType>
      <xs:restriction base="xs:string">
        <xs:enumeration value="add" />
        <xs:enumeration value="delete" />
        <xs:enumeration value="query" />
        <xs:enumeration value="set" />
      </xs:restriction>
    </xs:simpleType>
  </xs:attribute>
</xs:complexType>
<xs:complexType name="teammateType">
```

```
<xs:attribute name="id" type="xs:string" use="required"/>
</xs:complexType>
<xs:complexType name="playcollectRequestType">
  <xs:complexContent>
    <xs:extension base="base_requestType">
      <xs:sequence>
        <xs:element name="prompt" type="promptType"
          minOccurs="0"/>
        <xs:element name="pattern" type="dtmfGrammarType"
          minOccurs="0"/>
      </xs:sequence>
      <xs:attribute name="prompturl" type="xs:string"/>
      <xs:attribute name="offset" type="xs:string"/>
      <xs:attribute name="barge" type="yesnoType" default="yes"/>
      <xs:attribute name="promptencoding" type="xs:string"/>
      <xs:attribute name="cleardigits" type="yesnoType"
        default="no"/>
      <xs:attribute name="maxdigits" type="xs:string"/>
      <xs:attribute name="firstdigittimer" type="xs:string"
        default="5000ms"/>
      <xs:attribute name="interdigittimer" type="xs:string"
        default="2000ms"/>
      <xs:attribute name="extradigittimer" type="xs:string"
        default="1000ms"/>
      <xs:attribute name="interdigitcriticaltimer"
        type="xs:string"/>
      <xs:attribute name="skipinterval" type="xs:string"
        default="6s"/>
      <xs:attribute name="ffkey" type="DTMFkeyType"/>
      <xs:attribute name="rwkey" type="DTMFkeyType"/>
      <xs:attribute name="returnkey" type="DTMFkeyType"
        default="#" />
      <xs:attribute name="escapekey" type="DTMFkeyType"
        default="*" />
      <xs:attribute name="maskdigits" type="yesnoType"
        default="no"/>
    </xs:extension>
  </xs:complexContent>
</xs:complexType>
<xs:complexType name="playrecordRequestType">
  <xs:complexContent>
    <xs:extension base="base_requestType">
      <xs:sequence>
        <xs:element name="prompt" type="promptType"
          minOccurs="0"/>
      </xs:sequence>
      <xs:attribute name="prompturl" type="xs:string"/>
      <xs:attribute name="promptencoding" type="xs:string"/>
      <xs:attribute name="offset" type="xs:string" default="0"/>
      <xs:attribute name="barge" type="yesnoType" default="yes"/>
      <xs:attribute name="cleardigits" type="yesnoType"
        default="no"/>
    </xs:extension>
  </xs:complexContent>
</xs:complexType>
```

```
<xs:attribute name="escapekey" type="xs:string" default="*" />
<xs:attribute name="recurl" type="xs:string" use="required" />
<xs:attribute name="mode" default="overwrite">
  <xs:simpleType>
    <xs:restriction base="xs:NMTOKEN">
      <xs:enumeration value="append" />
      <xs:enumeration value="overwrite" />
    </xs:restriction>
  </xs:simpleType>
</xs:attribute>
<xs:attribute name="recencoding" type="xs:string" />
<xs:attribute name="initsilence" type="xs:string" />
<xs:attribute name="endsilence" type="xs:string" />
<xs:attribute name="duration" type="xs:string" />
<xs:attribute name="beep" type="yesnoType" default="yes" />
<xs:attribute name="recstopmask" type="xs:string"
  default="01234567890*#" />
</xs:extension>
</xs:complexContent>
</xs:complexType>
<xs:complexType name="managecontentRequestType">
  <xs:complexContent>
    <xs:extension base="base_requestType">
      <xs:attribute name="fetchtimeout" type="xs:string"
        default="10000" />
      <xs:attribute name="mimetype" type="xs:string" />
      <xs:attribute name="name" type="xs:string" />
      <xs:attribute name="httpmethod">
        <xs:simpleType>
          <xs:restriction base="xs:NMTOKEN">
            <xs:enumeration value="put" />
            <xs:enumeration value="post" />
          </xs:restriction>
        </xs:simpleType>
      </xs:attribute>
      <xs:attribute name="action">
        <xs:simpleType>
          <xs:restriction base="xs:NMTOKEN">
            <xs:enumeration value="move" />
            <xs:enumeration value="delete" />
          </xs:restriction>
        </xs:simpleType>
      </xs:attribute>
      <xs:attribute name="dest" type="xs:string" />
      <xs:attribute name="src" type="xs:string" use="required" />
    </xs:extension>
  </xs:complexContent>
</xs:complexType>
<xs:complexType name="stopRequestType">
  <xs:complexContent>
    <xs:extension base="base_requestType" />
  </xs:complexContent>
</xs:complexType>
```

```

</xs:complexType>
<xs:complexType name="faxRequestType">
  <xs:complexContent>
    <xs:extension base="base_requestType">
      <xs:sequence>
        <xs:element name="prompt" type="promptType" minOccurs="0"/>
      </xs:sequence>
      <xs:attribute name="lclid" type="xs:string"/>
      <xs:attribute name="prompturl" type="xs:string"/>
      <xs:attribute name="recurl" type="xs:string"/>
      <xs:attribute name="rmtid" type="xs:string"/>
    </xs:extension>
  </xs:complexContent>
</xs:complexType>
<xs:complexType name="dtmfGrammarType">
  <xs:choice>
    <xs:element name="regex" type="dtmfPatternType"
      maxOccurs="unbounded"/>
    <xs:element name="mgcpdigitmap" type="dtmfPatternType"/>
    <xs:element name="megacodigitmap" type="dtmfPatternType"/>
  </xs:choice>
</xs:complexType>
<xs:complexType name="dtmfPatternType">
  <xs:attribute name="value" type="xs:string" use="required"/>
  <xs:attribute name="name" type="xs:string"/>
</xs:complexType>
<!-- Definitions for base and concrete MSCML responses -->
<!-- and embedded types. -->
<xs:complexType name="base_responseType" abstract="true">
  <xs:attribute name="request" use="required">
    <xs:simpleType>
      <xs:restriction base="xs:NMTOKEN">
        <xs:enumeration value="configure_conference"/>
        <xs:enumeration value="configure_leg"/>
        <xs:enumeration value="play"/>
        <xs:enumeration value="playcollect"/>
        <xs:enumeration value="playrecord"/>
        <xs:enumeration value="managecontent"/>
        <xs:enumeration value="faxplay"/>
        <xs:enumeration value="faxrecord"/>
        <xs:enumeration value="stop"/>
      </xs:restriction>
    </xs:simpleType>
  </xs:attribute>
  <xs:attribute name="id" type="xs:string"/>
  <xs:attribute name="code" type="xs:string" use="required"/>
  <xs:attribute name="text" type="xs:string" use="required"/>
</xs:complexType>
<xs:complexType name="responseType">
  <xs:complexContent>
    <xs:extension base="base_responseType">
      <xs:sequence>

```

---

```

    <xs:element name="error_info"
      type="stoponerrorResponseType" minOccurs="0"/>
    <xs:element name="team" type="configure_teamResponseType"
      minOccurs="0"/>
  </xs:sequence>
  <xs:attribute name="reason" type="xs:string"/>
  <xs:attribute name="reclength" type="xs:string"/>
  <xs:attribute name="recduration" type="xs:string"/>
  <xs:attribute name="digits" type="xs:string"/>
  <xs:attribute name="name" type="xs:string"/>
  <xs:attribute name="playduration" type="xs:string"/>
  <xs:attribute name="playoffset" type="xs:string"/>
  <xs:attribute name="faxcode" type="xs:string"/>
  <xs:attribute name="pages_sent" type="xs:string"/>
  <xs:attribute name="pages_recv" type="xs:string"/>
</xs:extension>
</xs:complexContent>
</xs:complexType>
<xs:complexType name="stoponerrorResponseType">
  <xs:attribute name="code" type="xs:string" use="required"/>
  <xs:attribute name="text" type="xs:string" use="required"/>
  <xs:attribute name="context" type="xs:string" use="required"/>
</xs:complexType>
<xs:complexType name="configure_teamResponseType">
  <xs:sequence>
    <xs:element name="teammate" type="teammateType" minOccurs="0"
      maxOccurs="unbounded"/>
  </xs:sequence>
  <xs:attribute name="id" type="xs:string" use="required"/>
  <xs:attribute name="numteam" type="xs:integer" use="required"/>
</xs:complexType>

<!-- Definitions for MSCML event subscriptions and -->
<!-- embedded types -->
<xs:complexType name="conference_eventsubscriptionType">
  <xs:sequence>
    <xs:element name="events">
      <xs:complexType>
        <xs:sequence>
          <xs:element name="activetalkers"
            type="activetalkersSubscriptionType"/>
        </xs:sequence>
      </xs:complexType>
    </xs:element>
  </xs:sequence>
</xs:complexType>
<xs:complexType name="activetalkersSubscriptionType">
  <xs:attribute name="report" type="yesnoType" use="required"/>
  <xs:attribute name="interval" type="xs:string" default="60s"/>
</xs:complexType>
<xs:complexType name="leg_eventsubscriptionType">
  <xs:sequence>

```

```

    <xs:element name="events">
      <xs:complexType>
        <xs:sequence>
          <xs:element name="keypress"
            type="keypressSubscriptionType" minOccurs="0"
            maxOccurs="1"/>
          <xs:element name="signal" type="signalSubscriptionType"
            minOccurs="0" maxOccurs="unbounded"/>
        </xs:sequence>
      </xs:complexType>
    </xs:element>
  </xs:sequence>
</xs:complexType>
<xs:complexType name="keypressSubscriptionType">
  <xs:attribute name="report" use="required">
    <xs:simpleType>
      <xs:restriction base="xs:NMTOKEN">
        <xs:enumeration value="standard"/>
        <xs:enumeration value="long"/>
        <xs:enumeration value="both"/>
        <xs:enumeration value="none"/>
      </xs:restriction>
    </xs:simpleType>
  </xs:attribute>
  <xs:attribute name="maskdigits" type="yesnoType" default="no"/>
</xs:complexType>
<xs:complexType name="signalSubscriptionType">
  <xs:attribute name="type" type="xs:NMTOKEN" use="required"/>
  <xs:attribute name="report" type="yesnoType" use="required"/>
</xs:complexType>
<!-- Definitions for MSCML event notifications and -->
<!-- embedded types. -->
<xs:complexType name="conferenceNotificationType">
  <xs:sequence>
    <xs:element name="activetalkers"
      type="activetalkersNotificationType" minOccurs="0"/>
  </xs:sequence>
  <xs:attribute name="uniqueid" type="xs:string" use="required"/>
  <xs:attribute name="numtalkers" type="xs:string"
    use="required"/>
</xs:complexType>
<xs:complexType name="activetalkersNotificationType">
  <xs:sequence minOccurs="0">
    <xs:element name="talker" maxOccurs="unbounded">
      <xs:complexType>
        <xs:attribute name="callid" type="xs:string"
          use="required"/>
      </xs:complexType>
    </xs:element>
  </xs:sequence>
</xs:complexType>
<xs:complexType name="keypressNotificationType">

```

---

```

<xs:sequence>
  <xs:element name="status" type="statusType"/>
</xs:sequence>
<xs:attribute name="digit" type="DTMFkeyType" use="required"/>
<xs:attribute name="length" use="required">
  <xs:simpleType>
    <xs:restriction base="xs:NMTOKEN">
      <xs:enumeration value="standard"/>
      <xs:enumeration value="long"/>
    </xs:restriction>
  </xs:simpleType>
</xs:attribute>
<xs:attribute name="method" use="required">
  <xs:simpleType>
    <xs:restriction base="xs:NMTOKEN">
      <xs:enumeration value="standard"/>
      <xs:enumeration value="long"/>
      <xs:enumeration value="double"/>
    </xs:restriction>
  </xs:simpleType>
</xs:attribute>
<xs:attribute name="interdigittime" type="xs:string"
  use="required"/>
</xs:complexType>
<xs:complexType name="statusType">
  <xs:attribute name="command" use="required">
    <xs:simpleType>
      <xs:restriction base="xs:string">
        <xs:enumeration value="idle"/>
        <xs:enumeration value="play"/>
        <xs:enumeration value="collect"/>
        <xs:enumeration value="record"/>
      </xs:restriction>
    </xs:simpleType>
  </xs:attribute>
  <xs:attribute name="duration" type="xs:string" use="required"/>
</xs:complexType>
<xs:complexType name="signalNotificationType">
  <xs:attribute name="type" use="required" fixed="busy"/>
</xs:complexType>

<!-- Definitions for miscellaneous embedded, helper data types -->
<xs:complexType name="promptType">
  <xs:choice maxOccurs="unbounded">
    <xs:element name="audio" type="promptcontentType"/>
    <xs:element name="variable" type="spokenvariableType"/>
  </xs:choice>
  <xs:attribute name="locale" type="xs:string"/>
  <xs:attribute name="baseurl" type="xs:string"/>
  <xs:attribute name="stoponerror" type="yesnoType" default="no"/>
  <xs:attribute name="gain" type="xs:string" default="0"/>
  <xs:attribute name="gaindelta" type="xs:string" default="0"/>

```

```
<xs:attribute name="rate" type="xs:string" default="0"/>
<xs:attribute name="ratedelta" type="xs:string" default="0"/>
<xs:attribute name="repeat" type="xs:string" default="1"/>
<xs:attribute name="duration" type="xs:string"
  default="infinite"/>
<xs:attribute name="offset" type="xs:string" default="0"/>
<xs:attribute name="delay" type="xs:string" default="0"/>
</xs:complexType>
<xs:complexType name="promptcontentType">
  <xs:attribute name="url" type="xs:string" use="required"/>
  <xs:attribute name="encoding" type="xs:string"/>
  <xs:attribute name="gain" type="xs:string" default="0"/>
  <xs:attribute name="gaindelta" type="xs:string" default="0"/>
  <xs:attribute name="rate" type="xs:string" default="0"/>
  <xs:attribute name="ratedelta" type="xs:string" default="0"/>
</xs:complexType>
<xs:complexType name="spokenvariableType">
  <xs:attribute name="type" use="required">
    <xs:simpleType>
      <xs:restriction base="xs:NMTOKEN">
        <xs:enumeration value="dat"/>
        <xs:enumeration value="dig"/>
        <xs:enumeration value="dur"/>
        <xs:enumeration value="mth"/>
        <xs:enumeration value="mny"/>
        <xs:enumeration value="num"/>
        <xs:enumeration value="sil"/>
        <xs:enumeration value="str"/>
        <xs:enumeration value="tme"/>
        <xs:enumeration value="wkd"/>
      </xs:restriction>
    </xs:simpleType>
  </xs:attribute>
  <xs:attribute name="subtype">
    <xs:simpleType>
      <xs:restriction base="xs:NMTOKEN">
        <xs:enumeration value="mdy"/>
        <xs:enumeration value="dmy"/>
        <xs:enumeration value="ymd"/>
        <xs:enumeration value="ndn"/>
        <xs:enumeration value="t12"/>
        <xs:enumeration value="t24"/>
        <xs:enumeration value="USD"/>
        <xs:enumeration value="gen"/>
        <xs:enumeration value="ndn"/>
        <xs:enumeration value="crd"/>
        <xs:enumeration value="ord"/>
      </xs:restriction>
    </xs:simpleType>
  </xs:attribute>
  <xs:attribute name="value" type="xs:string" use="required"/>
</xs:complexType>
```



---

```
<xs:simpleType name="yesnoType">
  <xs:restriction base="xs:NMTOKEN">
    <xs:enumeration value="yes"/>
    <xs:enumeration value="no"/>
    <xs:enumeration value="1"/>
    <xs:enumeration value="0"/>
    <xs:enumeration value="true"/>
    <xs:enumeration value="false"/>
  </xs:restriction>
</xs:simpleType>
<xs:simpleType name="DTMFkeyType">
  <xs:restriction base="xs:string">
    <xs:pattern value="[0-9]"/>
    <xs:pattern value="[A-D]"/>
    <xs:pattern value="[a-d]"/>
    <xs:pattern value="#" />
    <xs:pattern value="\*" />
  </xs:restriction>
</xs:simpleType>
<xs:complexType name="gainType">
  <xs:choice>
    <xs:element name="auto" type="autogainType"/>
    <xs:element name="fixed" type="fixedgainType"/>
  </xs:choice>
</xs:complexType>
<xs:complexType name="autogainType">
  <xs:attribute name="startlevel" type="xs:string"/>
  <xs:attribute name="targetlevel" type="xs:string"/>
  <xs:attribute name="silencethreshold" type="xs:string"/>
</xs:complexType>
<xs:complexType name="fixedgainType">
  <xs:attribute name="level" type="xs:string"/>
</xs:complexType>
</xs:schema>
```

## D - RTP Codec Selection with VoiceXML <transfer/>

This appendix describes how the Real-Time Protocol (RTP) codec can work with the VoiceXML transfer element.

When <transfer/> is enabled, the IP Media Server sends a SIP INVITE with an SDP offer to the URI specified in the VoiceXML script. The contents of this offer, combined with capabilities of the called device, determine the type of media streams established. By default, the offer matches the capabilities of the existing, inbound call leg.

For example, if the inbound leg has both audio and video streams, the SDP offer might include media lines for both. In addition, customers have requested application-level control to override this standard behavior. With such control, the application can force an audio-only call even if the called device supports video.

To enable this control, the IP Media Server supports a proprietary VoiceXML attribute named `video` of the <transfer/> tag. This attribute has two supported values: `allow` and `suppress`.

Table 62 shows the media streams that are included in the SDP offer based on the negotiated SDP of the inbound call and the value of the `video` attribute. The default value of the `video` attribute is `allow`.

Table 62. SDP Offer Sent in Outbound SIP INVITE

Video Attribute Values			
		Allow Value	Suppress Value
Inbound Leg Negotiated SDP	Audio	Audio	Audio
	Audio+Video	Audio+Video	Audio

The called device on the outbound leg generates an SDP answer based on the IP Media Server's offer and its own capabilities. This answer determines the actual media streams that are created. So, the video attribute cannot guarantee that video media are used; only that the option is presented.

The IP Media Server also allows the client application to determine what media types were negotiated on the outbound call. The client application uses this function for billing purposes. A custom shadow variable to `<transfer/>` provides this information.

The IP Media Server standard supports:

```
<transfer name="name".../> shadow variables
    name$.duration, name$.inputmode and name$.utterance
```

The IP Media Server also supports the proprietary:

```
name$.media.
```

If the transfer did not connect to the far end, the value for the media shadow variable is "". If the media is connected with audio only, then the media shadow variable is set to audio. If the media is connected with both audio and video, then the shadow variable is set to audio+video.

## E - Using AMR-NB

The Dialogic® IP Media Server supports the following features of the Adaptive Multi-Rate - Narrow Band (AMR-NB) codec:

- ◆ Octet-alignment: **Bandwidth Efficient** (bit-aligned) or **Octet-aligned** (byte-aligned) Mode.
- ◆ Mode-set: 0 – 7 (bit rates from 4.75–12.2)
- ◆ Simple payload sorting.
- ◆ 1 channel per session.

The following features are not supported by the IP Media Server:

- ◆ Mode switching / CMR (Codec Mode Request).
- ◆ Unequal Bit-error Detection (UED) and Unequal Bit-error\_Protection (UEP), along with the associated checksums and or frame CRCs. (Refer to section 3.6 in rfc3267.)
- ◆ Forward Error Correction (FEC) or frame interleaving / redundant transmission. (Refer to section 3.7 in rfc3267.)
- ◆ Robust Sorting.
- ◆ Interleaving.

## Using AMR-NB

To use AMR-NB encoded media with your application, send a SIP INVITE request to the IP Media Server including an SDP section that describes the requested AMR-NB session.

The following example is a typical SDP description for an AMR-NB session:

```
m=audio 49120 RTP/AVP 97
a=rtpmap:97 AMR/8000
a=fmtp:97 mode-set=0,2,5,7; octet-align=1
a=maxptime:20
```

The component lines of an SDP description are described in the following sections.

---

Note: Only the first two lines (m= and a=rtpmap:) are required.

---

### Media Description Header

The “m=” media description header specifies the media name and transport address.

```
m=<media> <port> <transport> <fmt list>
```

Parameter	Description
media	“audio”
port	Port number
transport	“RTP/AVP”
fmt list	RTP dynamic payload type (number from 96–127)

For example, the line `m=audio 49120 RTP/AVP 97`, indicates the following:

- ◆ audio = audio media
- ◆ 49120 = using port number 49120
- ◆ RTP/AVP = IETF's Realtime Transport Protocol using the Audio/Video Profile carried over UDP
- ◆ 97 = RTP dynamic payload type ID

### Dynamic Payload Type

The “a= rtpmap:” attribute line defines a mapping between a codec and a dynamic payload type number. The following description for the “a= rtpmap:” attribute line shows how AMR-specific information is provided.

```
a=rtpmap:<payload type> <encoding name>/
<clock rate>[/<encoding parameters>]
```

Parameter	Description
payload type	RTP dynamic payload type specified in the m= line
encoding name	AMR
clock rate	8000
encoding parameters	Number of channels (Default:1)

For example, the attribute line `a=rtpmap:97 AMR/8000/1` indicates the following:

- ◆ 97= RTP dynamic payload type ID (97)
- ◆ AMR = Encoding type (name) is “AMR”.
- ◆ 8000 = Clock rate of 8000 samples per second
- ◆ 1 = Using 1 channel (optional value).

## Format Specific Parameters

The “a=ftmp:” attribute line specifies the characteristics of the AMR connection. The following description for the “a=ftmp:” attribute line shows how AMR-specific information is provided.

```
a=fmtp:<dpt> <format specific parameters>
```

Parameter	Description
dpt	RTP dynamic payload type (same value used in “m=” and “a=rtpmap:” lines)
format specific parameters	For AMR, a semi-colon separated list of <b>parameter=value</b> pairs (which may contain white space)

The AMR-specific parameters that can be received via the `a=fmtp:` line are described in the following table.

Parameter	Description						
octet-align= <b>X</b>	<p>Whether data is octet aligned..</p> <table border="1"> <thead> <tr> <th>Value</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>Octet-aligned (byte-aligned) operation.</td> </tr> <tr> <td>0 (default)</td> <td>Bandwidth efficient (bit-aligned) operation.</td> </tr> </tbody> </table>	Value	Description	1	Octet-aligned (byte-aligned) operation.	0 (default)	Bandwidth efficient (bit-aligned) operation.
Value	Description						
1	Octet-aligned (byte-aligned) operation.						
0 (default)	Bandwidth efficient (bit-aligned) operation.						
mode-set= <b>X</b>	<p>Requested set of AMR modes; restricts the set of active codec modes to a subset of all modes. Possible values are a comma-separated list of modes from the set: 0,...,7 (representing the 8 possible bit rates from lowest to highest). (Default: all.)</p> <p>The IP Media Server always employs the highest transmission rate available for sending AMR-encoded audio.</p>						
mode-change-period= <b>N</b>	<p>Interval (number of frames) at which codec mode changes are allowed. The initial phase of the interval is arbitrary, but changes must be separated by multiples of N frames. If this parameter is not present, mode changes are allowed at any time during the session.</p> <p><b>Note:</b> The IP Media Server never initiates a mode change.</p>						
mode-change-neighbor= <b>X</b>	<p>Whether mode changes are allowed to the neighboring modes in the active codec mode set. Neighboring modes are the ones closest in bit rate to the current mode, either the next higher or next lower rate.</p> <table border="1"> <thead> <tr> <th>Value</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>Mode changes are allowed only to the neighboring modes in the active codec mode set.</td> </tr> <tr> <td>0 (default)</td> <td>Change is allowed between any two modes in the active codec mode set.</td> </tr> </tbody> </table> <p><b>Note:</b> The IP Media Server never initiates a mode change.</p>	Value	Description	1	Mode changes are allowed only to the neighboring modes in the active codec mode set.	0 (default)	Change is allowed between any two modes in the active codec mode set.
Value	Description						
1	Mode changes are allowed only to the neighboring modes in the active codec mode set.						
0 (default)	Change is allowed between any two modes in the active codec mode set.						

Parameter	Description						
crc= <b>X</b>	<p>Whether CRCs are included in the payload.</p> <table border="1"> <thead> <tr> <th>Value</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>Frame CRCs may be included in the payload. crc=1 implies that octet-aligned operation must be used for the session.</td> </tr> <tr> <td>0 (default)</td> <td>Frame CRCs shall not be included in the payload.</td> </tr> </tbody> </table> <p><b>Note:</b> The IP Media Server does not support including CRSCs in the payload. The IP Media Server rejects the call if the far end (caller) specifies that CRC information is included in the AMR payload.</p>	Value	Description	1	Frame CRCs may be included in the payload. crc=1 implies that octet-aligned operation must be used for the session.	0 (default)	Frame CRCs shall not be included in the payload.
Value	Description						
1	Frame CRCs may be included in the payload. crc=1 implies that octet-aligned operation must be used for the session.						
0 (default)	Frame CRCs shall not be included in the payload.						
robust-sorting= <b>X</b>	<p>Whether the IP Media Server supports robust sorting.</p> <table border="1"> <thead> <tr> <th>Value</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>The payload uses robust sorting. robust-sorting=1 implies that octet-aligned operation must be used for the session.</td> </tr> <tr> <td>0 (default)</td> <td>The payload uses simple payload sorting.</td> </tr> </tbody> </table> <p><b>Note:</b> The IP Media Server does not support robust payload sorting. The IP Media Server rejects the call if the far end (caller) specifies robust sorting.</p>	Value	Description	1	The payload uses robust sorting. robust-sorting=1 implies that octet-aligned operation must be used for the session.	0 (default)	The payload uses simple payload sorting.
Value	Description						
1	The payload uses robust sorting. robust-sorting=1 implies that octet-aligned operation must be used for the session.						
0 (default)	The payload uses simple payload sorting.						
interleaving= ?	<b>Note:</b> The IP Media Server does not support interleaving. The IP Media Server rejects the call if the far end (caller) specifies interleaving.						
channels=	<p>Number of audio channels. (Default: 1.)</p> <p>The IP Media Server supports only a value of 1.</p>						

For example, the attribute line `a=fmtp:97 mode-set=0,2,5,7; octet-align=1` indicates the following:

- ◆ 97= RTP dynamic payload type ID (97)
- ◆ mode-set=0,2,5,7 = Available AMR modes are modes 0, 2, 5, and 7.
- ◆ octet-align=1 = AMR data is byte (octet) aligned.



## Optional Format Parameters

Additional optional attribute lines may also be included in the SDP section, for example:

### Packet Time

```
a=ptime:<milliseconds>
```

Length of time represented by the media in a packet, in milliseconds. This is probably only meaningful for audio data. It should not be necessary to know ptime to decode RTP audio, and it is intended as a recommendation for the encoding / packetization of audio.

### Maximum Packet Time

```
a=maxptime:20
```

Maximum amount of media that can be encapsulated in a payload packet, expressed as time in milliseconds. The time is calculated as the total time represented by the media present in the packet. The time should be a multiple of the frame size. If this parameter is not present, the sender MAY encapsulate a maximum of 16 encoded frames into one RTP packet. This is a media attribute, and is not dependent on charset.

Values supported by the IP Media Server: 20 (the desired value), 40 (the maximum value).

# F - Dial Pulse Detection

This appendix describes the Dial Pulse Detection feature and includes the following sections:

- ◆ [Overview](#)
- ◆ [Configuration](#)

---

## Overview

In addition to the existing checks for Dual Tone Multi Frequency (DTMF) and Call Progress Analysis (CPA) tones, the IP Media Server can be configured to detect Dial Pulses.

### Dial Pulse and DTMF

This feature supports all of the features associated with the DTMF signal detection. It is designed to minimize the considerations that an application developer has to address when supporting both DTMF and Dial Pulse detection.

The following applies to Dial Pulse detection and DTMF detection:

- ◆ Dial Pulse detector can detect pulse dialed digits in any call state that allows DTMF detection.
- ◆ Dial Pulse detection runs concurrently with DTMF detection. You can configure the Dial Pulse detector to disable itself when it first receives a DTMF digit. This option is configured by the parameter `DP_detectAfterDtmf` (See Table on page 388.)
- ◆ The `mserv` process detects Dial Pulse and DTMF within a single stream. As a result, when `mserv` detects each type of signaling, it checks for the completion of any previously detected digits of the alternate type.

### Consistency

In order for `mserv` to detect Dial Pulse digits, they must be fairly consistent in form and meet the criteria defined in the configuration parameters. You can manipulate these parameters to meet the dial pulse criteria for differing networks and implementations.

### MSCML and VXML 1.0/2.0 Support

Dial Pulse detection is compatible with MSCML and VXML applications - both VXML versions 1.0 and 2.0.

When collecting dial pulse digits under VXML, the following VXML properties can be set so collection will not time out. Values will depend on the application and average dial pulse timeout.

```
<property name="timeout" value="10s"/>
<property name="interdigittimeout" value="10s"/>
```

The default values of these properties for VXML 1.0 and 2.0 are different. If the interdigit timing is large, VXML may declare a timeout while collecting digits. VXML 2.0 is more sensitive to this issue as its default values are smaller. Use the VXML code provided to change these default values.

## Configuration

Dial Pulse detection is configured separately from DTMF detection. DTMF is always detected and Dial Pulse detection is configurable.

Dial Pulse is configured with the `DP_DetectAfterDtmf` parameter. If that parameter is set to Y, the IP Media Server detects Dial Pulse after detecting a DTMF. If configured to N, the IP Media Server no longer detects Dial Pulse after detecting a DTMF.

### Parameters

The `mserv` process detects the properties of a dial pulse such as those represented by the parameters in the following table. You can configure these parameters only in the `snowshore.cfg` file (keyword=value).

Table 63. Dial Pulse Parameters

Parameter	Description	Default (in Decimal)	Range
<code>DP_enabled</code>	DP Enabled Determines if the Dial Pulse detection is enabled or disabled system-wide	N	Y or N
<code>DP_detectAfterDtmf</code>	DP Detect After Dtmf Determines if Dial Pulses should be detected on a call leg after DTMF has been detected on that leg.	N	Y or N
<code>DP_low_Thresh</code>	Low Threshold Level Sample level required to be below in order to declare sample as a Low Signal	5,000	0 to 32768
<code>DP_lowThreshCnt</code>	Low Threshold Count Number of consecutive Low samples to be considered a post Dial Pulse Low (or trough)	17	0 to 32768
<code>DP_peakThresh</code>	Peak Threshold Sample value required for signal to be declared a dial pulse peak	14,000	20 to 32768

Parameter	Description	Default (in Decimal)	Range
DP_maxTransCnt	Max Transition Count Maximum low to high transition time (samples). For a peak to be declared, the ramp time has to be less than this number of samples	30,000	0 to 32768
DP_settleByCnt	Settle By Count Number of consecutive samples that the signal must be below the threshold to declare the signal as a dial pulse.	400	0 to 65535
DG_InterDigitQuietCnt	Inter-Digit Quiet Count Number of 10 ms frames between the last pulse of a previous digit and the first pulse of the next digit.	30	20 to 32768

The following figure illustrates these parameters and the description follows.

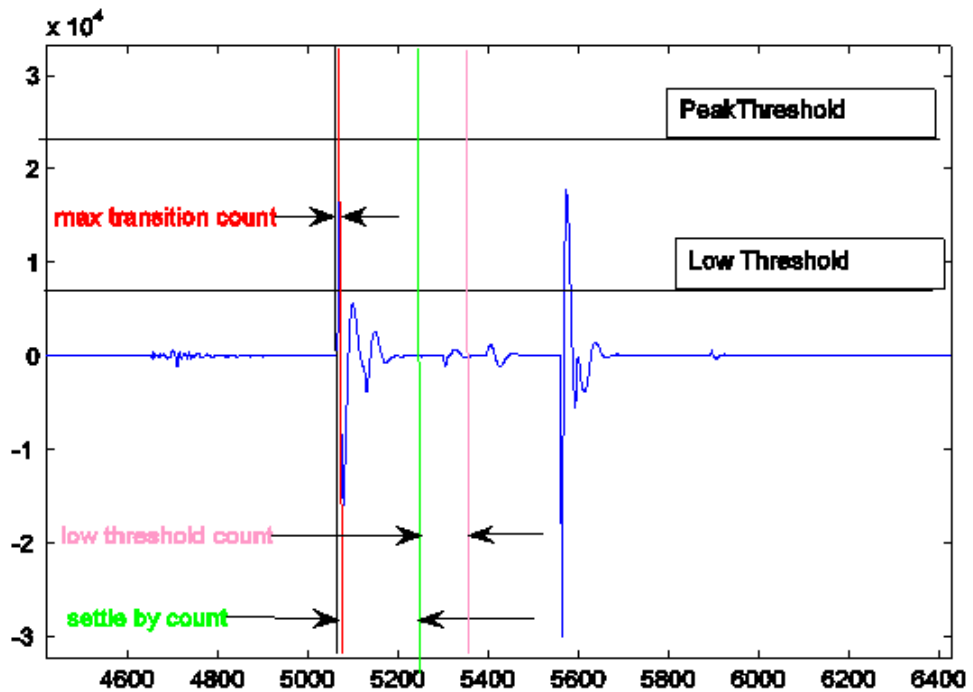


Figure 18. Dial Pulse Parameters

### Description of Dial Pulse Algorithm

The first representative pulse is a digit 1. Note the pair of pulses. Each of the configurable parameters is indicated by a line on the figure. The second pulse is ignored by the algorithm because it does not exceed the peak threshold at any time.

The first pulse exceeds the Peak Threshold of approximately 25,000 within a small enough number of samples (defined by Max Trans Count) to constitute the start of Dial Pulse.

The first pulse then returns below the Low Threshold and remains there (settles) between the red and green vertical lines.

The number of packets received during this interval is the Settle By Count. This point is the end of a detected dial pulse.

Between the green and pink lines, mserv counts the consecutive samples the signal remains below the Low Threshold. This information is useful when determining the time spacing between pulses used to determine if two consecutive pulses are from the same digit or from different digits.

# G - MSCML MRCP Session Management

This appendix describes MRCP Session Management and includes the following sections:

- ◆ [Overview of MRCP Session Management](#)
- ◆ [MSCML Requests](#)
- ◆ [Sample Call Flow](#)

# Overview of MRCP Session Management

Applications that include Automated Speech Recognition (ASR) and Text-to-Speech (TTS) features are most commonly written in VoiceXML. However, certain applications may require ASR and TTS capabilities which are not exposed through existing versions of the VoiceXML standard. Further, some developers may not wish to write or rewrite their applications using VoiceXML.

To address these situations the IP Media Server supports MSCML extensions to create and terminate MRCP v2 sessions on behalf of the application.

The application is responsible for implementing all required MRCP client functions. The IP Media Server's role is simply to manage the MRCP session and associated media streams.

---

**Note:** MRCP Session Management is supported with the following packages: Nuance OSR MRCP 3.0.10 and SWMS 4.0 running under Linux.

---

## Features Enabled

MRCP Session Management enables the following features:

- ◆ speech synthesis
- ◆ speech recognition
- ◆ speech recording

## Process

Figure 19 outlines the MRCP Session Management process. Each stage is explained in the table following the figure.



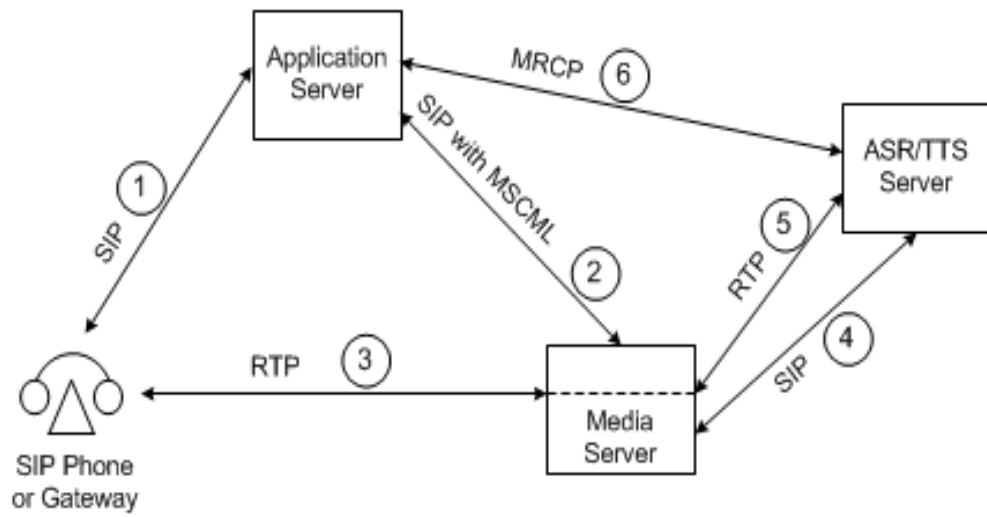


Figure 19. MRCP Session Management

Table 64. Description of MRCP Session Management Topology

Stage	Description
1	SIP session between a caller and the application server that invokes the application.
2	SIP session between the application server and IP Media Server used to set up RTP communication between the caller and IP Media Server and to control media processing. The application server uses this session to send MSCML <mrcp_session_create> and <mrcp_session_terminate> requests.
3	RTP session with caller. This is normally full duplex, as shown.
4	SIP session between the IP Media Server and the ASR/TTS server, which supports MRCP V2.
5	RTP session between the IP Media Server and the ASR/TTS server. This is normally half-duplex, since ASR and TTS rarely reside on the same server. In this example, media is sent from the IP Media Server to an ASR server, which performs speech recognition.
6	MRCP session between the application server and the ASR/TTS server. The application server uses the information contained in the IP Media Server's response to the <mrcp_session_create> request to establish this connection. Once the connection is established, the application server communicates directly with the ASR/TTS server using MRCP and starts a recognition session.

---

## MSCML Requests

The following MSCML requests support MRCP Session Management. The formats and attributes of each are provided below.

- ◆ `mrcp_session_create`
- ◆ `mrcp_session_terminate`

These MSCML requests are scoped to the SIP dialog in which the requests are sent and apply to the dialog's associated RTP streams. Any usage of MRCP resources will create at least one new SIP dialog with the desired MRCP resource.

If multiple resource types are supported on a single MRCP server, the media server may reuse the existing dialog and simply send a re-INVITE with the SDP that describes the additional resource desired. However, different resources such as ASR and TTS are often deployed on separate servers for performance reasons so a separate SIP dialog will get established for each MRCP resource type in this configuration. The SDP for that dialog contains a single resource type.

The lifetimes of any SIP dialogs with MRCP resources are tied to the original SIP dialog with the application server. The media server must clean up any MRCP related SIP dialogs when the original SIP dialog is terminated. Standard session timers apply to MRCP dialogs.

Established MRCP sessions can only be ended by the `<mrcp_session_terminate/>` request or termination of the parent SIP dialog as described above. Unlike the `<play>`, `<playrecord>` and `<playcollect>` requests, the MRCP sessions are not stopped when another MSCML request is received.

### Create session request format

The following is the MSCML format of the `mrcp_session_create` request.

```
<?xml version="1.0" encoding="UTF-8"?>
  <MediaServerControl version="1.0">
    <mrcp_session_create id="1234"
      resource="speechrecog"
      url="sip:mrcpserver_address" proto="TCP/MRCPv2"
      rtpdirection="sendonly"/>
  </MediaServerControl>
```

## Attributes

The following are the attributes of the `mrcp_create_session` request.

Table 65. Attributes of `mrcp_create_session`

Attribute	Mandatory	Description
<code>id</code>	No	An application-provided transaction identifier.
<code>resource</code>	Yes	The resource to be provided by the MRCP server
<code>url</code>	Yes	The complete SIP URL of the MRCP server. Note that in actual deployments this will most likely be a SIP proxy, which will select a specific server and forward the request.
<code>proto</code>	No	The transport protocol to be used for the MRCP session. Alternatives are "TCP/MRCPv2" (default) and "TCP/TLS/MRCPv2". This value is treated as case-sensitive in SDP and will be treated likewise here.
<code>rtpdirection</code>	No	The value to be used in the <code>direction</code> attribute of the SDP offer created by the media server. The attribute is interpreted from the perspective of the media server. This value is treated as case-sensitive in SDP and will be treated likewise here.  If this attribute is not set the SDP offer will be based on the resource required. For example the RTP stream to a "speechrecog" resource is send-only from the perspective of the media server.

## Create session response format

The following is the MSCML format of the `mrcp_session_response` request.

```
<?xml version="1.0" encoding="UTF-8"?>
  <MediaServerControl version="1.0">
    <response id="1234" request="mrcp_session_create"
      code="200" text="OK"
      connection="192.168.16.2" port="32416"
      proto="TCP/MRCPv2"
      channel="32AECB234338@speechrecog" />
  </MediaServerControl>
```

Table 66. Attributes of mrcp\_session\_response

Attribute	Required	Description
id	No	An application-provided transaction identifier.
connection	Yes	The IP address of the MRCP server the client connects to.
port	Yes	The port of the MRCP server that the client connects to.
proto	Yes	The transport protocol to be used for the MRCP session. Alternatives are "TCP/MRCPv2" and "TCP/TLS/MRCPv2".
connectiontype	Yes	The value of the "a=connection" attribute associated with the MRCP control channel from the MRCP server's SDP answer. Alternatives are "new" and "existing." There is no MSCML default value - the MRCP server is required to provide this information in its SDP answer.
channel	Yes	The channel identifier returned from the MRCP server. The client uses it to communicate with the server.

## Terminate session request format

```
<?xml version="1.0" encoding="UTF-8"?>
  <MediaServerControl version="1.0">
    <mrcp_session_terminate id="1234"
      channel="32AECB234338@speechrecog"/>
  </MediaServerControl>
```

Table 67. Attributes of mrcp\_terminate\_session\_request

Attribute	Required	Description
id	No	An application-provided transaction identifier.
channel	Yes	The channel identifier returned from the MRCP server. The client uses it to communicate with the server.

## Terminate session response format

```
<?xml version="1.0" encoding="UTF-8"?>
  <MediaServerControl version="1.0">
    <response id="1234" request="mrcp_session_terminate"
      code="200" text="OK"/>
  </MediaServerControl>
```

An `<error_info>` element will be included in the response if the MRCP server returns an error. This enables the MRCP server error code and text, as well as the target URL that generated the error, to be returned to the application.

# Sample Call Flow

This section provides a sample call flow of the following MRCP session creation scenario.

## Scenario

- ◆ After creating a SIP dialog with the media server the application establishes an MRCP session with a speech recognition resource and starts a recognition session via MSCML requests to the IP Media Server.
- ◆ The application then requests the media server to begin playing a prompt which is barged by speech input from the user. The MRCP server sends the results of the speech recognition to the application.
- ◆ When finished the application terminates the MRCP session and media server session.

The following sequence of interactions assume that the initial dialog between the application server and media server is already established.

## Call Flow

### 1 Create MRCP Session

The application sends an MSCML request to create an MRCP session on an existing dialog using a SIP INFO message. The MSCML request must specify the type of resource needed and must include the URL of the MRCP server to contact. The request may optionally include an MSCML request ID for the purpose of matching the request with its response.

The follow is an example of the MSCML payload:

```
<?xml version="1.0" encoding="UTF-8"?>
  <MediaServerControl version="1.0">
    <mrcp_session_create id="1234"
      resource="speechrecog"
      url="sip:speechresources@server.example.com"
      proto=TCP/MRCPv2"
      rtpdirection="sendonly"/>
  </MediaServerControl>
```

- a. As a result, the media server generates a SIP INVITE to the target address as shown below.

```
INVITE sip:speechresources@server.example.com SIP/2.0
Via:SIP/2.0/TCP ssmediaserver.example.com:5060;
branch=z9hG4bK74bf9
Max-Forwards:6
To:sip:speechresources@server.example.com
From:SnowShoreMediaServer
<sip:ssms@example.com>;tag=1928301774
```

```
Call-ID:a84b4c76e66710
CSeq:314161 INVITE
Contact:<sip:ssms@example.com>
Content-Type:application/sdp
Content-Length: 230
```

```
v=0
o=SnowShoreMediaServer 2890844526 2890842808
IN IP4 126.16.64.4
s=-
c=IN IP4 224.2.17.12
m=application 9 TCP/MRCPv2
a=setup:active
a=connection:new
a=resource:speechrecog
a=cmid:1
m=audio 49170 RTP/AVP 0 101
a=rtpmap:0 pcmu/8000
a=sendonly
a=mid:1
```

- b. The speech server generates a response.

```
SIP/2.0 200 OK
Via:SIP/2.0/TCP ssmediaserver.example.com:5060;
branch=z9hG4bK74bf9
```

```
To:sip:speechresources@server.example.com;tag=a6c85cf
From:SnowShoreMediaServer
<sip:ssms@example.com>;tag=1928301774
Call-ID:a84b4c76e66710
CSeq:314161 INVITE
Contact:<sip:ssms@example.com>
Content-Type:application/sdp
Content-Length: 249
```

```
v=0
o=speechresources 2890844526 2890842808 IN IP4
126.16.64.4
s=-
c=IN IP4 224.2.17.12
m=application 32416 TCP/MRCPv2
a=setup:passive
a=connection:new
a=channel:32AECB234338@speechrecog
a=cmid:1
m=audio 48260 RTP/AVP 00 101
a=rtpmap:0 pcmu/8000
a=recvonly
a=mid:1
```

- c. The media server acknowledges the response.



```

ACK sip:mresources@server.example.com SIP/2.0
Via:SIP/2.0/TCP ssmediaserver.example.com:5060;
branch=z9hG4bK74bf9
Max-Forwards:6
To:sip:speechresources@server.example.com;tag=a6c85cf

From:SnowShoreMediaServer
<sip:ssms@example.com>;tag=1928301774
Call-ID:a84b4c76e66710
CSeq:314162 ACK
Content-Length:0

```

1. The media server sends an MSCML response to the <mrpc\_session\_create> request, as depicted in the following example.

```

<?xml version="1.0" encoding="UTF-8"?>
<MediaServerControl version="1.0">
  <response id="1234" request="mrpc_session_create"
    code="200" text="OK" connect="224.2.17.2"
    port="32416" channel="32AECB234338@speechrecog"/>
</MediaServerControl>

```

---

**Note:** The IP Media Server's response must contain the connection and port information for the control channel and the channel identifier returned from the MRCP server. These are required to establish the control channel and identify all future requests for the associated resource.

---

2. The application initiates the start speech recognition by making an MRCP RECOGNIZE request on the control channel with the correct channel id and grammar to match. Following is an example request:

```

MRCP/2.0 487 RECOGNIZE 32416
Channel-Identifier:32AECB23433801@speechrecog
Confidence-Threshold:0.9
Content-Type:application/srgs+xml
Content-Id:<request1@form-level.store>
Content-Length:104

<?xml version="1.0"?>

<!-- the default grammar language is US English -->
<grammar xmlns="http://www.w3.org/2001/06/grammar"
  xml:lang="en-US" version="1.0" root="request">

  <!-- single language attachment to tokens -->
  <rule id="yes">
    <one-of>
      <item xml:lang="fr-CA">oui</item>

```

```

        <item xml:lang="en-US">yes</item>
    </one-of>
</rule>

<!-- single language attachment to a rule expansion -->
<rule id="request">may I speak to
    <one-of xml:lang="fr-CA">
        <item>Alex Michaels</item>
        <item>Bill Fuller</item>
    </one-of>
</rule>
</grammar>

```

- The MRCP server responds that recognition is in progress.

```

MRCP/2.0 48 32614 200 IN-PROGRESS
Channel-Identifier:32AECB23433801@speechrecog

```

- The application requests the media server to play a sequence of prompts by sending a request like the following.

```

<MediaServerControl version="1.0">
  <request>
    <play id="33298">
      <prompt stoponerror="yes"
        baseurl="file:///var/mediaserver/prompts/"
        <audio url="prompt1.wav"/>
        <audio url="prompt2.wav"/>
        <audio url="prompt3.wav"/>
      </prompt>
    </play>
  </request>
</MediaServerControl>

```

- The MRCP server sends a response when speech input has started.

```

MRCP/2.0 49 START-OF-INPUT 32614 IN-PROGRESS
Channel-Identifier:32AECB23433801@speechrecog

```

- The application tells the media server to terminate prompt play to implement barge-in based on speech input. The MSCML request would look like:

```

<?xml version="1.0" encoding="UTF-8"?>
<MediaServerControl version="1.0">
  <request>
    <stop id="5678"/>
  </request>
</MediaServerControl>

```

- The IP Media Server sends a response to the <stop> request.

```

<?xml version="1.0" encoding="UTF-8"?>
<MediaServerControl version="1.0">
  <response id="5678" request="stop" code="200"
    text="OK"/>
</MediaServerControl>

```

8. The IP Media Server sends a response for the stopped <play> request similar to the following:

```
<?xml version="1.0" encoding="UTF-8"?>
<MediaServerControl version="1.0">
  <response id="33298" request="play" code="200"
    text="OK" reason="stopped" playduration="12620">
  </response>
</MediaServerControl>
```

9. The MRCP server sends a response when recognition has completed with the results of the recognition. The response would look like the following:

```
MRCP/2.0 465 RECOGNITION-COMplete 32614 COMPLETE
Channel-Identifier:32AECB23433801@speechrecog
Completion-Cause:000 success
Waveform-
URI:<http://web.media.com/session123/audio.wav>;
size=342456;duration=25435
Content-Type:application/nlsml+xml
Content-Length:276
```

```
<?xml version="1.0"?>
<result grammar="session:request1@form-level.store">
  <interpretation>
    <instance name="Person">
      <Person>
        <Name> Alex Michaels</Name>
      </Person>
    </instance>
    <input> may I speak to Alex Michaels</input>
  </interpretation>
</result>
```

10. The application issues an MRCP GET-RESULT request to recompute and retrieve the results. Following is an example request.

```
MRCP/2.0 73 GET-RESULT 32614
Channel-Identifier:32AECB23433801@speechrecog
Confidence-Threshold:0.9
```

11. The MRCP server responds to the GET-RESULT request with a message similar to the one below.

```
MRCP/2.0 487 32614 200 COMPLETE
Channel-Identifier:32AECB23433801@speechrecog
Content-Type:application/nlsml+xml
Content-Length:276
```

```
<?xml version="1.0"?>
<result grammar="session:request1@form-level.store">
  <interpretation>
    <instance name="Person">
      <Person>
        <Name> Alex Michaels </Name>
      </Person>
    </instance>
  </interpretation>
</result>
```

```
        </instance>
        <input> may I speak to Alex Michaels </input>
    </interpretation>
</result
```

12. The application sends a BYE to the media server which causes the media server to also issue a BYE to the MRCP server. The result is that resources on both servers are freed.

## H - T.30 Fax Detection

The T.30 Fax Detection feature enables the IP Media Server to detect T.30 Fax CNG tones embedded in a G.711 RTP audio stream, and to notify a VXML application script via a VXML event. The VXML script then transfers the media stream to a target Fax machine for further processing.

The transfer is "hairpinned", and does not go through the IP Media Server conference mixer, which in turn keeps the audio (fax) signals cleaner and does not introduce unnecessary delay.

---

**Note:** These enhancements are implemented only in the VoiceXML browser.

---

# Fax Call Transfer Call Flow

Figure 20 outlines the call flow between the initiating (sending) Fax machine and the terminating (receiving) Fax machine. Each step is explained in the [Description](#) that follows the figure.

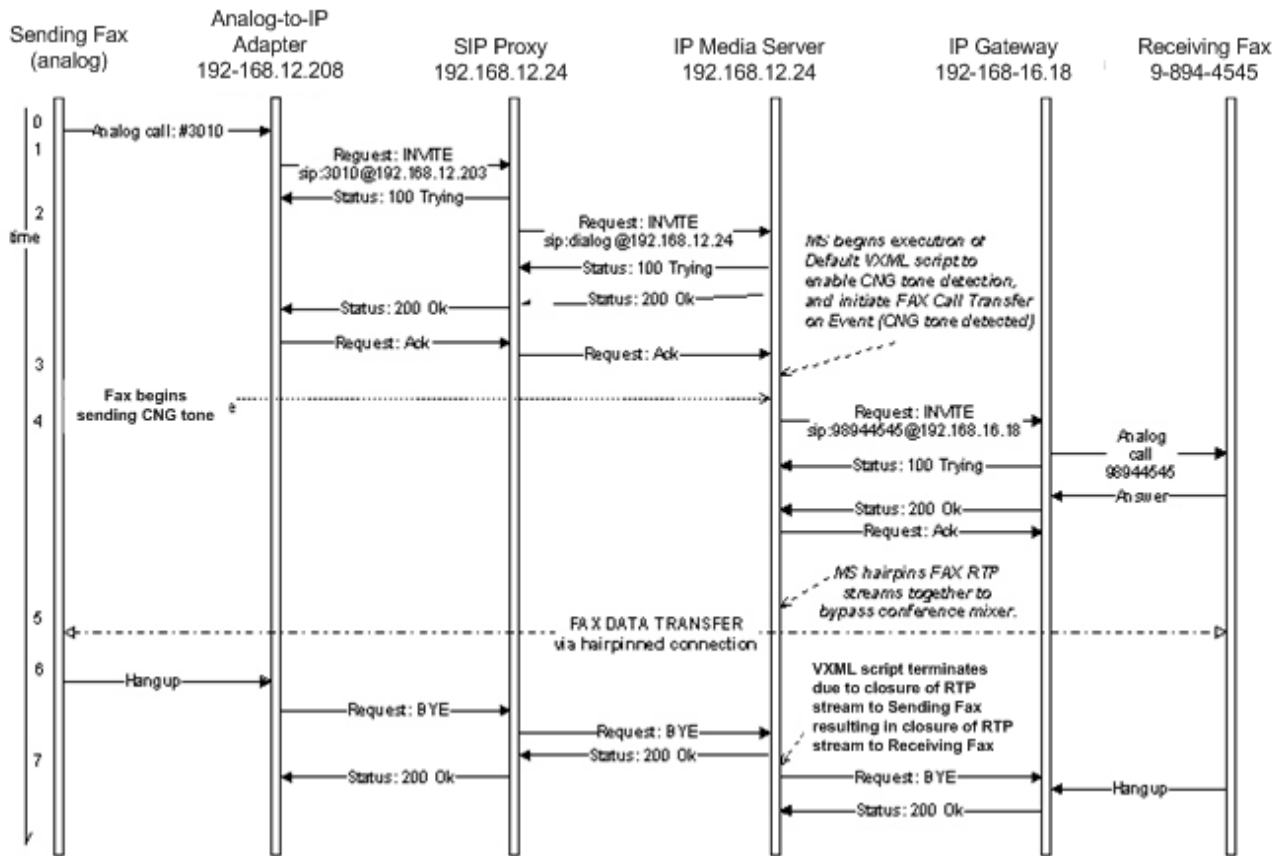


Figure 20. Fax Call Transfer

## Description

The following contains a detailed description of the call flow for the Fax Call Transfer scenario.

Time	Description
1	The Sending Fax machine initiates an analog call (dialed number is 3010) to the Analog-to-IP Adapter.
2	The SIP proxy server has been configured to respond to the INVITE 3010 by issuing a SIP INVITE request to the IP Media Server. Note that the INVITE request specifies using the dialog service.
3	A connection is established between the Sending Fax machine and the IP Media Server. The IP Media Server begins executing the default VXML script, which has requested the CNG tone detection. The Sending Fax machine is generating the CNG tone.
4	<p>The CNG tone is detected by the mserv process and a notification event (CNG) is sent to the VXML script. In response to this event, the VXML script performs a Fax call &lt;transfer&gt; specifying bridge="forward")</p> <p>This step results in an INVITE 98944545@192.168.16.18 sent to the IP Gateway. The IP Gateway calls the Receiving Fax (via the PBX) at the number 9-894-4545. When the call is established, the VXML script instructs the mserv process to hairpin together the Receiving Fax and the Sending Fax RTP streams.</p>
5	The Sending Fax machine and the Receiving Fax server are connected together via hairpinning RTP streams. The two Fax machines continue to perform the fax data transfer.
6	The Sending Fax completes sending the Fax to the Receiving Fax and hangs up. The Analog-to-IP Adapter issues a SIP BYE request to the SIP Proxy which results in the SIP Proxy sending a BYE request to the IP Media Server.
7	The VXML script terminates when the Sending Fax RTP stream closes which closes the Receiving Fax stream.

## CNG Tone Detection and Event Notification

The VXML interpreter supports the ability to enable and disable CNG tone detection, and to handle event notification (CNG-tone-detected events). The VXML application (or VXML script) employs this capability to handle incoming Fax calls.

### VoiceXML Implementation

The property "***com.snowshore.signal***" enables detection of various signals on the IP Media Server platform when the property is in scope in the VoiceXML interpreter.

---

**Note:** The property has a value string that allows the scriptwriter to specify which signal or signals that should be detected. The only signal type implemented at this time is Fax CNG tone. For example,

```
<property name="com.snowshore.signal" value="CNG" />
```

enables detection of CNG tone. Multiple signals could be enabled by placing them all in the "value" attribute, separated by commas.

---

When the CNG tone signal is detected by the IP Media Server platform, the VoiceXML interpreter throws the "***com.snowshore.signal***" event. The VXML script should handle this event and then disable signal detection. The "\_message" variable is set to the type of signal, and can be detected as shown in the following example:

```
<catch event="com.snowshore.signal">
  <if cond="_message == 'CNG' ">
    <!--Perform Fax Call Transfer when CNG tone is detected -->
  ->
</if>
```

Note that `<catch event=>` does not block; events are handled in an asynchronous manner, which permits other operations to take place.

In order to disable CNG tone detection, the application must clear the "***com.snowshore.signal***" property in the correct scope. For example:

```
<property name="com.snowshore.signal" value="OFF" />
```

The following list describes the interaction of CNG tone detection with other VXML input collection modes:

- ◆ If there is a DTMF recognition session, it is terminated and any digits collected up to that point are dropped.
- ◆ If a prompt that allows barge-in is being played, it is stopped.



The following list describes the VXML interpreter's actions when it is not in an input collection mode:

- ◆ If the prompt does not allow barge-in, the VXML interpreter processes the event after the prompt completes.
- ◆ If the VXML interpreter is in a transition state that does not require the collection of input, the event is left queued until the VXML interpreter enters the input state.

## VoiceXML Properties

### **com.snowshore.signal**

Enables detection of various signals on the IP Media Server platform when the property is in scope in the VoiceXML interpreter.

---

**Note:** The property has a value string that allows the scriptwriter to specify which signal or signals that should be detected. The only signal type implemented at this time is Fax CNG tone. For example,

```
<property name="com.snowshore.signal" value="CNG" />
```

enables detection of CNG tone. Multiple signals could be enabled by placing them all in the "value" attribute, separated by commas.

---

When the CNG tone signal is detected by the IP Media Server platform, the VoiceXML interpreter throws the "**com.snowshore.signal**" event. The VXML script should handle this event and then disable signal detection. The "\_message" variable is set to the type of signal, and can be detected as shown in the following example:

```
<catch event="com.snowshore.signal">
  <if cond="_message == 'CNG'">
    <!--Perform Fax Call Transfer when CNG tone is detected -->
  ->
  </if>
```

Note that `<catch event=>` does not block; events are handled in an asynchronous manner, which permits other operations to take place.

## Fax Detection—Example Script

The following example illustrates the syntax to enable CNG tone detection and to catch the tone-detected event.

### Welcome.vxml:

```
<?xml version="2.0" encoding="UTF-8"?>

<vxml version="2.0">

<property name="universals" value="help"/>
<property name="timeout" value="3000"/>
<property name="com.snowshore.signal" value="CNG"/>

<form id="f1">

  <field name="F_1">
    <dtmf> [1-3]
    </dtmf>
    <prompt bargein="false">
      <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/msg040"/>
    </prompt>

    <help count="1" cond="dtmf">
      <prompt> since the condition is set to false, this
handler will be skipped
      <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/msg060"/>
    </prompt>
    </help>

    <noinput count="1" cond="true">
      <prompt> this is the first occurrence of the help
event. say help again.</prompt>
    </noinput>

    <noinput count="2" cond="true">
      <prompt> this is the second occurrence of the help
event. Do something!!!.</prompt>
    </noinput>

    <noinput count="3" cond="true">
      <prompt> this is the third occurrence of the help
event. goodbye.</prompt>
      <disconnect/>
    </noinput>

    <catch event="com.snowshore.signal">
```

```
<prompt> GOT THE com.snowshore.signal
event.</prompt>
  <if cond="_message=='CNG'">
    <prompt> Found CNG message. GOTO fax transfer
support</prompt>
    <goto next='transfer_call.xml' />
  </if>
  <disconnect />
  <prompt> goodbye.</prompt>
</catch>

</field>

<filled>
  <prompt>why did you fill the field?</prompt>
</filled>

</form>
```

## Fax Call Transfer

The VXML script can perform a call transfer (using the VXML <transfer> element). The transfer initiates a call to the Fax termination server (to which the Fax is sent). When the call is successful (i.e., connection has been established between the IP Media Server and the Fax termination server), the RTP streams are joined to a conference.

The VoiceXML specification defines the <transfer> element for establishing a second call leg and allowing it to interact with the initial leg. The IP Media Server extends the <transfer> element by adding a new value “forward” to the bridge attribute. The forward bridge mode causes the two RTP streams to be “hairpinned” together. Hairpinning prevents the audio data from being mixed within the conference mixer.

---

**Note:** Only two RTP streams can be hairpinned together, and they must employ the same encoding type (otherwise end-to-end communication is not possible).

---

### Call Transfer—Example Script

The following example script illustrates the use of Fax Call Transfer (i.e., hairpinning the RTP streams together) and call termination.

#### Transfer\_call.xml:

```
<?xml version="2.0"?>
<!-- 10.10.10.111 is the SnowShore MPS -->
<!-- 1234567@192.169.12.207 is a valid SIP destination-
-->
<!-- of the receiving Fax server. -->
<!-- 1234567 is associated with techsupport@cantata.com
-->

<!-- This script first plays "star", followed by "One"
-->
<!-- Then attempts to do the transfer. If the transfer
is -->
<!-- success, when the call is ended, you'll hear the
reason-->
<!-- number (see below in the script) followed by the
number of-->
<!-- seconds the call is connected between the caller
and -->
<!-- the callee. -->

<vxml version="2.0">
<form name="genericTransfer">
  <block>
```

```
<audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/star.neu" />
</block>
<var name="fax2emailURI"
expr="session.telephone.dnis + '@192.168.12.207'"/>
<var name="subscriber" expr="'sip:' +
session.telephone.dnis + '@cantata.com'"/>
<transfer name="makethecall" destexpr="subscriber"
requiexpr="fax2emailURI" connecttimeout="10s"
bridge="forward">
  <prompt>
    <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/1.ris" />
    </prompt>
    <filled>
      <if cond="makethecall == 'busy'">
        <prompt>
          <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/2.ris" />
          </prompt>
          <elseif cond="makethecall == 'noanswer'"/>
            <prompt>
              <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/3.ris" />
              </prompt>
              <elseif cond="makethecall ==
'near_end_disconnect'"/>
                <prompt>
                  <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/4.ris" />
                  </prompt>
                  <elseif cond="makethecall ==
'far_end_disconnect'"/>
                    <prompt>
                      <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/5.ris" />
                      </prompt>
                      <elseif cond="makethecall ==
'network_disconnect'"/>
                        <prompt>
                          <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/6.ris" />
                          </prompt>
                          <elseif cond="makethecall == 'network_busy'"/>
                            <prompt>
```

```
        <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/7.ris" />
        </prompt>
        <elseif />
        <prompt>
            <value class="number"
expr="makethecall$.duration" />
            <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/seconds.neu" />
            </prompt>
        </if>
    </filled>
</transfer>
<block>
    <audio
src="http://localhost/snowshore/prompts/gmvoices/en
_US/susan/star.neu" />
</block>
</form>
</vxml>
```

# I - T.38 Fax Detection, Termination, and Initiation

This chapter describes the T.38 Fax Detection, Termination, and Initiation feature.

The fax termination server is incorporated into the IP Media Server using the Dialogic® Brooktrout® T.38 host fax termination software. Refer to the *Dialogic® IP Media Server Installation and Operations Guide* for instructions to configure the fax software.

## Detection and Termination

This feature enables the Dialogic® IP Media Server to detect CNG Fax Tones embedded in a G.711 RTP audio stream, and notify a VXML application script via a VXML event. The IP Media Server then initiates a SIP Re-INVITE method to the endpoint requesting a switch from the current voice coder mode to T.38 fax relay mode. Upon the completion of the SIP Re-INVITE method, the Dialogic® IP Media Server will record the fax transmission to a TIFF file stored locally on the IP Media Server and optionally convert the file to PDF format.

## Initiation (Email to Fax)

This feature enables the IP Media Server to initiate a T.38 Fax Transmission of a file received as an attachment in an SMTP request.

Users can send an email to the IP Media Server in the following format:

```
user@phonenumber . fax
```

Once the IP Media Server receives the email, it parses the email into components. The IP Media Server can parse the following formats and convert them to TIFF format:

- ◆ JPG
- ◆ GIF
- ◆ TIFF
- ◆ PDF
- ◆ Text

Once the IP Media Server converts the email to TIFF format, it saves it with a unique filename in the /tmp directory.

The IP Media Server notifies the T.38 application about the new fax file to transmit. The T.38 application accepts the destination phone number and the filename as input. The T.38 API transmits the fax and returns a status once the transmission is complete.



---

## Snowshore.cfg Parameters

You must configure the following parameters in the snowshore.cfg file to enable this feature.

Table 68. T.38 Fax Parameters

Parameter	Description	Default in decimal	Range
SR140APP_PORT	First port for the T.38 fax server	9000	9000-40000
SR140APP_PROXY_PORT	Host name for proxy fax SIP Server	5060	
SR140APP_PROXY_IP	Required field for sending fax	No default	

# Fax Call Termination Call Flow

Figure 21 outlines the call flow between the initiating (sending) fax server and the terminating (receiving) fax server. Each step is explained in the [Description](#) that follows the figure.

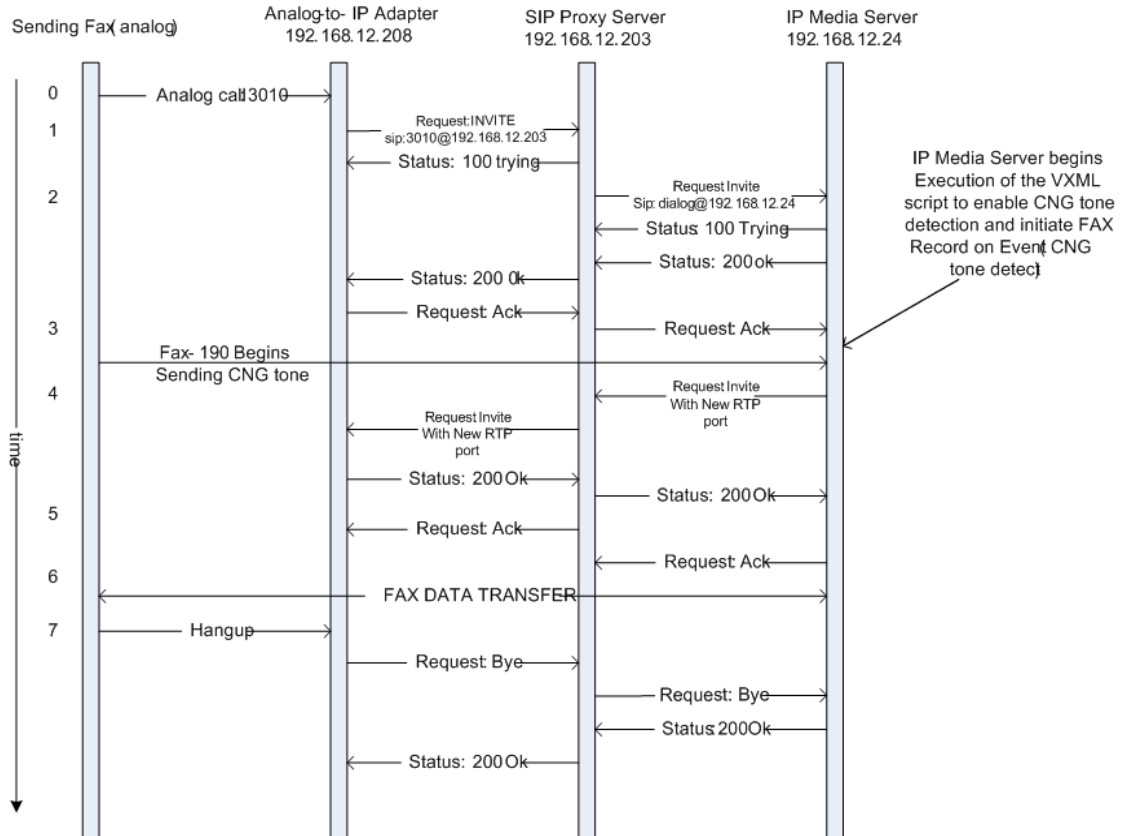


Figure 21. Fax Call Transfer

## Description

The following is a detailed description of the call flow for the Fax Call Transfer scenario.

Time	Description
1	The Sending Fax server initiates an analog call (dialed number is 3010) to the Analog-to-IP Adapter.
2	The Analog-to IP Adapter accepts the analog call and issues a SIP INVITE request to the SIP Server it is configured to use (for example, the SIP Proxy Server @192.168.12.203).
3	The SIP Proxy Server is configured to respond to the INVITE 3010 by issuing a SIP INVITE request to the IP Media Server. The INVITE request specifies the use of the dialog service.
4	A connection is established between the Sending Fax server and the IP Media Server. The Media Server begins execution of the default VXML script which has requested CNG tone detection. The Sending Fax server generates a CNG tone.
5	The CNG tone is detected by the mserv application and a notification event (CNG detected) is sent to the VXML script. In response to the event, the VXML script performs a fax record <record> (specifying dest_enc="tiff"). This results in a RE-INVITE request sent to 192.168.12.203 with the new RTP port for the fax data.
6	The Sending Fax server starts sending fax data to the T.38 Host Server RTP port to be terminated. The two fax systems continue to perform the fax data transfer.
7	The VXML script sends the fax tiff file to a file server.

## CNG Tone Detection and Event Notification

The VXML 1.0 interpreter supports the ability to enable and disable CNG tone detection, and to handle event notification (CNG-tone-detected events). The VXML application (or VXML script) employs this capability to handle incoming Fax calls.

The mserv application enables and disables the CNG tone detection through the report Media Server Protocol (MSP) command's Call Progress Analysis parameters. In addition, the mserv application provides event notification back to the VXML 1.0 browser / application.

### VoiceXML 1.0 Implementation

The property *"com.snowshore.signal"* enables detection of various signals on the IP Media Server platform when the property is in scope in the VoiceXML interpreter.

---

**Note:** The property has a value string that allows the scriptwriter to specify which signal or signals that should be detected. The only signal type implemented in this release is Fax CNG tone. For example:

```
<property name="com.snowshore.signal" value="CNG" />
```

This example enables detection of CNG tone. Multiple signals could be enabled by placing them all in the "value" attribute, separated by commas.

---

When the CNG tone signal is detected by the IP Media Server platform, the VoiceXML interpreter throws the *"com.snowshore.signal"* event. The VXML script should handle this event and then disable signal detection. The *"\_message"* variable is set to the type of signal, and can be detected as shown in the following example:

```
<catch event="com.snowshore.signal">
  <if cond="_message == 'CNG' ">
    <!--Perform Fax Call Transfer when CNG tone is detected -->
  ->
</if>
```

Note that `<catch event=>` does not block; events are handled in an asynchronous manner, which permits other operations to take place.

In order to disable CNG tone detection, the application must clear the *"com.snowshore.signal"* property in the correct scope. For example,

```
<property name="com.snowshore.signal" value="OFF" />
```

The following list describes the interaction of CNG tone detection with other VXML input collection modes:

- ◆ If there is a DTMF recognition session, it is terminated and any digits collected up to that point are dropped.
- ◆ If a prompt that allows barge-in is being played, it is stopped.

The following list describes the VXML interpreter's actions when it is not in an input collection mode:

- ◆ If the prompt does not allow barge-in, the VXML interpreter processes the event after the prompt completes.
- ◆ If the VXML interpreter is in a transition state that does not require the collection of input, the event is left queued until the VXML interpreter enters the input state.

## Fax Detection—Example VXML Script

The example below shows the syntax to enable CNG tone detection and catch the event.

```
<form id="f1">
  <field name="field1">
    <property name="com.snowshore.signal" value="CNG"/>
    <noinput>
      <!-- sets the value of this field, so that the FIA
      doesn't select this again -->
      <assign name="field1" expr="true"/>
    </noinput>
    <catch event="com.snowshore.signal">
      <if cond="_message == 'CNG'">
        <!-- do stuff when CNG is detected -->
      </if>
    </catch>
    <!-- This field waits for the CNG tone -->
    ....
  </field>
  <field name="Voicemail"/>
    <!-- When this field is visited, CNG is not enabled
    (the property is not set) -->
    <prompt> Welcome to VoiceMail. To leave a message
    ... </prompt>
    ....
  </field>
</form>
```

## VXML Fax Record

The VXML script can perform a fax record (using the VXML <record> element with the mimetype "image/tiff"). The record will initiate the media sever to redirect the incoming media to the T.38 fax termination server (the fax will be recorded as a tiff file).

The Fax Record feature is supported by the VoiceXML 1.0 browser through enhancements to the <record> element mimetype "image/tiff" and/or "image/pdf".

## Call Record - Example VXML Script

This section contains a sample VoiceXML 1.0 FAX Call Record script. The script demonstrates use of FAX Call Record and HTTP post of the file.

```
<vxml version="1.0">
  <var name="recordedMedia"/>

  <form id="recordMessage">
    <record name="deposit" beep="true" type="image/tiff"
      >
      <filled>
        <assign name="recordedMedia" expr="deposit"/>
        <goto next=" #deliverMessage"/>
      </filled>
    </record>
  </form>

  <form id="deliverMessage">
    <block>
      <submit method="post" namelist="recordedMedia"
        next="http://localhost/snowshore/htdocs/depositfax.
        cgi?ani='session.telephone.ani'"/>
    </block>
  </form>
```

# Index

## Symbols

# key 159  
\* key 159  
= 184, 322  
? 184, 322

## Numerics

100rel 86  
2833 events 109  
3GPP 36, 37, 38, 228, 229  
3GPP2 36, 37, 38, 228, 229

## A

ACK method (SIP) 62  
active talker events 33, 114, 120, 123  
activetalkers (MSCML) 145  
advanced conferences  
  active talker events 123  
  audio 129  
  control leg 119, 124  
  creating 119  
  defaults 115  
  DTMF 127  
  joining participants 121, 258  
  modifying conference 120, 257  
  modifying participants 122  
  MSCML 134  
  parking participants 124, 262  
  playing to 119, 124  
  recording 119, 124  
  removing participants 122  
Advanced conferencing attributes  
  reserveconf media 115  
  reserved talkers 115  
annc 31, 92, 96, 97  
announcement parameters 97

  delay 101  
  duration 102  
  early 101  
  play 40, 100  
announcement sequences 41, 95  
announcement service indicator 31  
  *See also* annc  
announcement variables  
  digits 103  
  money 103  
  months 104  
  numbers 104  
  silence 104  
  strings 104  
  time 104  
  weekdays 105  
announcements 40, 95  
  default locations 41  
  in conferences 150  
  Request-URI 96  
  sequences 95, 98  
  simple 41, 95  
  storage 40  
  types of 95  
application (VoiceXML) 348  
application servers 30, 55  
assign element (VoiceXML) 191, 326  
attribute lines (SDP) 84  
Attributes  
  endsilence 162  
  initsilence 162  
  playrecord 161, 162  
  stop\_on\_error 162  
attributes  
  stop\_on\_error 163  
audio 36  
  DTMF digits 39, 147  
  duration of play 102  
  encoding 174  
  file retrieval 95  
  in conferences 124, 129, 148  
  mixing 148  
  playing 97  
  recording 161, 166  
  supported CODECs 83  
  voice encoding 36

audio element (VoiceXML) 191, 326

audio files

- cardinal numbers 309
- Chinese Mandarin 306
- dates 310
- default location 41, 95
- in Sound Library 40
- letters of the alphabet 311
- ordinal numbers 310
- press key phrases 313
- standard phrases 307
- storage 40

## B

barge (MSCML) 158, 159, 161, 168

bargein (VoiceXML) 348

bargeintype (VoiceXML) 348

base (VoiceXML) 348

baseurl (MSCML) 168

beep (VoiceXML) 349

block element (VoiceXML) 192, 327

break element (VoiceXML) 327

bridge (VoiceXML) 349

bridge attribute 412

## C

cached fonts 238

call control 30

Call-ID header (SIP) 70

channels 42

choice element (VoiceXML) 193, 198, 328

class (VoiceXML) 349

clear element (VoiceXML) 194, 329

cleardigits (MSCML) 168

CLI

configuration for conferences 107

CNG tones 405

code (MSCML) 180

CODECs 79, 83

cond (VoiceXML) 350

conf= 31, 106

conference control leg 119, 124

conference participants

- advanced conferences 121
- audio mixing 148
- joining to advanced conference 121
- joining to simple conference 108
- modifying 122
- playing audio to 129
- removing 122
- simple conferences 109

conference service indicator 106

conferences 42

advanced 114

audio 129

control leg 114

creating advanced 119

creating simple 108

defaults for advanced 115

defaults for simple 109

DTMF digits 127, 147

ending simple 108

examples 108

joining participants 108, 121

mixing 42

modifying advanced 120, 122

parking participants 124

playing announcements in 150

recording 124, 166

removing participants 122

simple 42, 108

conferencing attributes

activetalkers (MSCML) 145

configure\_team (MSCML) 145

dtmfclamp (MSCML) 146, 147

events (MSCML) 146

id (MSCML) 148

keypress (MSCML) 146

mixmode 148

mixmode (MSCML) 148

notification (MSCML) 146

report (MSCML) 150

reserveconfmedia 150

reserveconfmedia (MSCML) 150

reservedtalkers 151

signal 146

subscribe 146

subscribe (MSCML) 146

toneclamp (MSCML) 151

type 151

type (MSCML) 151

configure\_conference 114

configure\_conference (MSCML) 114, 120, 134

configure\_leg 114, 139

configure\_leg (MSCML) 114, 121, 136, 137

configure\_team 139

configure\_team (MSCML) 145

conformance

HTTP 38

NFS 38

connection information (SDP) 83

Contact header (SIP) 70

content attribute (VoiceXML) 350

content element (VoiceXML) 195

content types (SIP) 55

Content-Length header (SIP) 70

Content-Type header (SIP) 71

control agent 30

control leg (advanced conferences) 114, 119, 124

count attribute (VoiceXML) 350

creating



advanced conferences 119  
 simple conferences 108  
 CSeq header (SIP) 71

## D

data element (VoiceXML) 196, 329  
 default application 31  
 default location  
   simple announcements 96  
   SnowShore prompts 95  
 delay (announcement parameter) 101  
 delay (MSCML) 169  
 dest attribute (VoiceXML) 351  
 dialog 31, 33  
 dialog service indicator 183, 321  
 Digit Buffering 158  
 digit buffering (MSCML) 158  
 digit collection (MSCML) 158  
 digits 179  
   barging (MSCML) 159, 161  
   buffering 158  
   collection 127, 165, 170, 171  
   ending collection 176  
   flushing 168  
   maximum collected 173  
 digits (MSCML) 179  
 digits (variables) 103  
 direction (SDP) 84  
 disconnect element (VoiceXML) 197, 330  
 double keypress 128  
 DTMF 39, 147, 158, 165  
   barging 168  
   detecting in conferences 127  
   escape key 170  
   flushing digits 168  
   return key 176  
   VoiceXML grammar 187, 324  
 DTMF attribute (VoiceXML) 351  
 dtmf element (VoiceXML) 330  
 dtmfclamp (MSCML) 116, 121, 137, 145, 146, 147  
 dtmfterm attribute (VoiceXML) 351  
 duration 169  
 duration (MSCML) 169  
 duration= (announcement parameter) 102  
 dynamic fonts 237

## E

early= (announcement parameter) 77, 101  
 ECMAScripting 230, 321  
 else element (VoiceXML) 197, 331  
 elseif element (VoiceXML) 197, 331  
 encoding (MSCML) 169  
 ending

simple conferences 108  
 endpoint devices 30  
 endsilence (MSCML) 169, 170  
 equals sign 184, 322  
 error element (VoiceXML) 198, 331  
 escapekey 159  
 escapekey (MSCML) 170  
 escaping 184, 322  
 event attribute (VoiceXML) 352  
 events 33, 123, 127  
 events (MSCML) 146  
 examples  
   IVR with MSCML 282  
   late-media announcements 247  
   MSCML 266  
   MSCML multipart syntax 90  
   VoiceXML 295  
 exit element (VoiceXML) 198, 332  
 expr (VoiceXML) 352  
 expritem (VoiceXML) 352  
 extradigittimer (MSCML) 170

## F

FAX detect 405  
 field element (VoiceXML) 199, 332  
 filled element (VoiceXML) 200, 333  
 finalsilence (VoiceXML) 352  
 firstdigittimer (MSCML) 170  
 font element (VoiceXML) 201  
 font files 237  
 form element (VoiceXML) 202, 333  
 forward value (bridge attribute) 412  
 From header (SIP) 71

## G

G.711 83  
 gain control 300  
 goto element (VoiceXML) 202, 334  
 grammar element (VoiceXML) 203, 334

## H

hairpinning 405, 412  
 headers  
   SDP 79  
 help element (VoiceXML) 205, 334  
 Hold Behavior 92  
 Hold media 83, 91, 155  
 Hold SDP 92  
 HTTP 38, 41, 95  
 http-equiv (VoiceXML) 352

**I**

id (MSCML) 148, 179  
id (VoiceXML) 352  
INFO method (SIP) 66, 154  
    and MSCML 32  
initsilence (MSCML) 171  
interdigittimer (MSCML) 171  
INVITE method (SIP) 60  
    and MSCML 32

**IVR**

    digit collection 165  
    escape key 170  
    functions 154  
    return key 176  
ivr 31, 32, 93, 154  
IVR attributes  
    barge 168  
    baseurl 168  
    beep 168  
    cleardigits 168  
    delay 169  
    duration 169  
    encoding 169  
    endsilence 169, 170, 178  
    escapekey 170  
    extradigittimer 170  
    firstdigittimer 170  
    id 171  
    initsilence 171, 175  
    interdigittimer 171  
    maxdigits 173  
    mode 173  
    offset 173  
    promptencoding 174  
    recencoding 174  
    recstopmask 174  
    recurl 174  
    repeat 149, 175  
    returnkey 176  
    stop\_on\_error 176  
    subtype 176  
    type 177  
    url 178

**IVR responses**

    code 180  
    digits 180  
    error\_info 179  
    id 179  
    playduration 180  
    reason 180  
    reclength 181  
    text 181

IVR service indicator 153

**J**

joining participants

    advanced conferences 121  
    simple conferences 108

**K**

keypress (MSCML) 146

**L**

locale (MSCML) 171

**M**

Maskdigits 128  
maskdigits 171  
maxdigits (MSCML) 173  
maxtime (VoiceXML) 353  
Media Content Recovery 189, 196, 324, 325, 329  
media information (SDP) 82  
Media Server Control Markup Language 32  
method (VoiceXML) 353  
methods (SIP) 60  
mid-call requests 66, 154  
MIME 57, 114, 228, 229  
mixed audio/video 37  
mixing (in conferences) 42, 148  
mixmode (MSCML) 116, 121, 124, 148  
modal (VoiceXML) 353  
mode (VoiceXML) 354  
modifying advanced conferences 120, 257  
money (PDML) 103  
months (PCML) 104  
MSCML 32, 33, 106, 169, 179  
    activetalkers 145  
    advanced conferences 42  
    barge 168  
    baseurl 168  
    cleardigits 168  
    code 180  
    conferencing attributes 145, 146, 147  
    conferencing dtmfclamp 147  
    conferencing events 146  
    conferencing id 148  
    conferencing keypress 146  
    conferencing mixmode 148  
    conferencing notification 146  
    conferencing report 150  
    conferencing requests 134  
    conferencing reserveconfmedia 150  
    conferencing subscribe 146  
    conferencing toneclamp 151  
    conferencing type 151  
    configure\_team 145  
    delay 169  
    digit collection 158  
    dtmfclamp 146, 147  
    encoding 169

- endsilence 169, 170
  - escapekey 170
  - events 146
  - extradigittimer 170
  - firstdigittimer 170
  - id 148, 179
  - initsilence 171
  - interdigittimer 171
  - keypress 146
  - maxdigits 173
  - MIME type 114
  - mixmode 148
  - multipart syntax 90
  - notification 146
  - prompt 166
  - promptencoding 174
  - recencoding 174
  - reclength 181
  - recstopmask 174
  - report 150
  - reserveconfmedia 150
  - reservedtalkers 151
  - responses 33, 157, 160, 164
  - responses for playrecord 163
  - sample code 266
  - signal 146
  - subscribe 146
  - subtype 176
  - syntax 90
  - text 181
  - toneclamp 151
  - type 151, 152, 177
  - value 178
- MSCML directives for IVR functions
- play 155
  - playcollect 155
  - playrecord 155
  - stop 155
- MSCML locale 171
- msecs (VoiceXML) 354
- msVXMLCriticalError 208, 337
- msVXMLLastCriticalError 208, 337
- multimedia files 37
- ## N
- name (VoiceXML) 354
  - namelist (VoiceXML) 354
  - next (VoiceXML) 355
  - nextitem (VoiceXML) 355
  - NFS 41, 95
  - notification (MSCML) 146
  - numbers (variables) 104
- ## O
- overlay element (VoiceXML) 211
- ## P
- packet time (SDP) 84
  - PDF 26
  - play (MSCML) 124, 129
    - sample code 282
  - Play and/or Collect 158
  - play= 40
  - play= (announcement parameter) 40, 100
  - playcollect (MSCML) 124, 127
    - sample code 266, 284
  - playrecord (MSCML) 124, 125, 161
    - sample code 266, 284
  - Playrecord attributes
    - recurl 161
  - playrecord attributes
    - beep 162
    - duration 162
    - mode 162
    - recencoding 162
  - port 5060 85, 95
  - prompt (MSCML) 166
  - promptencoding (MSCML) 174
  - provisional responses 86
  - ptime (SDP) 84
- ## Q
- question mark 184, 322
- ## R
- Real Time Streaming Protocol (RTSP) 38, 48
    - VCR controls 300
    - VCR controls example 302
  - recencoding (MSCML) 174
  - reclength (MSCML) 181
  - recording 161, 169, 171
    - encoding 174
    - length of 181
    - mode 173
    - MSCML request for 166
    - terminating 174
    - URL for 174
  - Record-Route header (SIP) 72
  - recsrc (VoiceXML) 355
  - recstopmask (MSCML) 174
  - recvonly (SDP) 84
  - re-INVITE (SIP) 61
  - repeat= (announcement parameter) 101
  - report (MSCML) 150
  - report values
    - both 128
    - long 128
    - none 128

- standard 128
- Requests (SIP) 58
- Request-URI (SIP) 58, 96, 174
  - parameters 100
- reserveconfmedia (MSCML) 115, 120, 124, 150
- reservedtalkers 115, 151
- reservedtalkers (MSCML) 120, 151
- responses
  - for MSCML 178
  - SIP 75
- Return codes (SIP) 75
- returnkey 159
- returnkey (MSCML) 176
- RFC conformance 51
- root document 185, 323
- Route header (SIP) 72

## S

- scope (VoiceXML) 355
- scroll element (VoiceXML) 218
- SDP 32, 61, 62, 79
  - attribute lines 84
  - connection information 83
  - direction 84
  - headers 79
  - Hold media 83, 91, 155
  - media description headers 81
  - media information 82
  - optional headers 84
  - packet time 84
  - session description headers 79
  - syntax errors 79
  - time description headers 81
  - TTL 83
- sendonly (SDP) 84
- sendrecv (SDP) 84
- sequences 41, 98
  - See also* announcement sequences
- service indicators 31, 56
  - annc 31
  - announcements 31, 96
  - conf 106
  - conf= 31
  - conference 106
  - default 31
  - dialog 31
  - ivr 31, 155
- session 31
- Session Description Protocol 79
  - See also* SDP
- session timer 61, 72
- Session-Expires header (SIP) 72
- signal 146
- signal (MSCML) 146
- silence 169
  - and recording 169, 171
  - as announcement variables 104
- simple announcements 41, 95
  - default location 96
  - overview 97
- Simple conference attributes
  - dtmfclamp 109
  - mixmode 109
  - type 109
- simple conferences 42, 108
  - creating 108
  - default parameters for 109
  - ending 108
  - joining participants to 108
  - upgrading 108
- SIP 30
  - announcement parameters 100
  - default port 85
  - event notification 33, 123
  - message body 55
  - ports 95
  - provisional responses 86
  - Requests 58
  - Request-URI 58, 174
  - responses 75
  - session description 62
  - syntax 55
  - transaction 56
- SIP headers 68
  - Call-ID 70, 123
  - compact form 69
  - Contact 70
  - Content-Length 70
  - Content-Type 71
  - CSeq 71
  - From 71
  - Record-Route 72
  - Route 72
  - Session-Expires 72
  - Supported 73
  - To 73
  - Unsupported 74
  - Via 74
- SIP methods 60
  - ACK 62
  - INFO 32, 66, 114, 124, 135, 154
  - INVITE 32, 60
- SIP Return Codes 75
- size (VoiceXML) 355
- slot (VoiceXML) 356
- softswitch 30
- src (VoiceXML) 356
- Star and Pound Keys 159
- stop (MSCML) 164
  - sample code 285
- stopdigits (VoiceXML) 356
- strings (PDML) 104
- subdialog element (VoiceXML) 219, 342

subscribe 123  
 subscribe (MSCML) 123, 135, 146  
 subscribe element 115  
 subtype (MSCML) 176  
 Supported header (SIP) 73  
 syntax overview  
   SDP 79  
   SIP 55

## T

text (MSCML) 181  
 text overlay 47, 195, 236  
 throw element (VoiceXML) 220, 343  
 time (PDML) 104  
 timeout (VoiceXML) 357  
 timing attributes 162  
   extradigittimer 159  
   firstdigittimer 159  
   interdigittimer 159  
 To header (SIP) 73  
 toneclamp (MSCML) 116, 121, 137, 151  
 tones 39  
 transfer element 412  
 transfer element (VoiceXML) 222, 344  
 trcr.cfg 201, 236, 238, 240  
 TTL 83  
 type (MSCML) 116, 121, 137, 151, 152, 177  
 type (VoiceXML) 357

## U

Unsupported header (SIP) 74  
 URIs  
   recurl attribute 174

## V

value (MSCML) 178  
 value (VoiceXML) 357, 358  
 value element (VoiceXML) 225, 346  
 valuetype (VoiceXML) 358  
 var element (VoiceXML) 225, 346  
 variable subtypes (PDML) 102  
 variable types (PDML) 102  
 VCR (VoiceXML) 358  
 VCR controls 300  
 version (VoiceXML) 358  
 Via header (SIP) 74  
 video 37  
 video transcoder 45  
   auto setting rules 46  
   restrictions 46  
   settings 45

video transcoding 45  
 VoiceXML 33  
   cond 350  
   elements 191, 326  
   grammars supported 187, 324  
   sample code 295  
 VoiceXML attributes  
   application 348  
   bargain 348  
   bargointype 348  
   base 348  
   beep 349  
   bridge 349  
   class 349  
   content 350  
   count 350  
   dest 351  
   dtmf 351  
   dtmfterm 351  
   event 352  
   expr 352  
   expritem 352  
   finalsilence 352  
   http-equiv 352  
   id 352  
   maxtime 353  
   method 353  
   modal 353  
   mode 354  
   msecs 354  
   name 354  
   namelist 354  
   next 355  
   nextitem 355  
   recsrc 355  
   scope 355  
   size 355  
   slot 356  
   src 356  
   stopdigits 356  
   timeout 357  
   type 357  
   value 357, 358  
   valuetype 358  
 VCR 358  
   version 358  
 VoiceXML elements  
   assign 191, 326  
   audio 191, 326  
   block 192, 327  
   break 327  
   choice 193, 198, 328  
   clear 194, 329  
   content 195  
   disconnect 197, 330  
   dtmf 330  
   else 197, 331  
   elseif 197, 331  
   error 198, 331  
   exit 198, 332  
   field 199, 332

filled 200, 333  
font 201  
form 202, 333  
goto 202, 334  
grammar 203, 334  
help 205, 334  
overlay 211  
scroll 218  
subdialog 219, 342  
throw 220, 343  
transfer 222, 344, 412  
value 225, 346  
var 225, 346  
vxml 226, 347  
VoiceXML interpreter 183, 321  
vxml element (VoiceXML) 226, 347

## W

weekdays (variables) 105