

IOS语音XML网关到CVP呼叫流使用MRCPv2 ASR/TTS

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简介

语音扩展标记语言(VXML)是万维网联盟定义的标准(W3C)。它设计创建提供被综合的语音，所说的话识别，DTMF位的识别的音频对话和记录的发言的音频。VXML服务器和客户端使用著名的HTTP协议交换VXML文档/页。

思科语音门户(CVP)提供可以在电话访问的智能和交互语音应答(IVR)应用程序。有三种CVP配置类型：

1. 独立服务
2. CVP呼叫控制
3. 呼叫队列和转移

Text-to-Speech (TTS)和自动语音识别服务器提供被综合的语音和所说的话/DTMF位功能的识别(ASR)。IOS® VXML网关与TTS/ASR服务器联络通过梅迪亚资源控制协议(MRCP)。有两个版本MRCP (RFC 4463)即，MRCPv1 (在RTSP的MRCP)和MRCPv2 (在SIP的MRCP)。

本文描述IOS语音XML网关的呼叫流对在使用MRCPv2 TTS/ASR服务器的独立服务部署的CVP呼叫。示例药房应用程序被实施了在CVP VXML服务器。

先决条件

要求

本文档没有任何特定的要求。

使用的组件

本文档中的信息基于以下软件和硬件版本：

- IOS VXML网关：思科AS5400XM，IOS 12.4(15)T1
- VXML服务器：CVP 4.0
- ASR/TTS服务器：Loquendo语音套件7.0

本文档中的信息都是基于特定实验室环境中的设备编写的。本文档中使用的所有设备最初均采用原始（默认）配置。如果您使用的是真实网络，请确保您已经了解所有命令的潜在影响。

规则

有关文档规则的详细信息，请参阅 [Cisco 技术提示规则](#)。

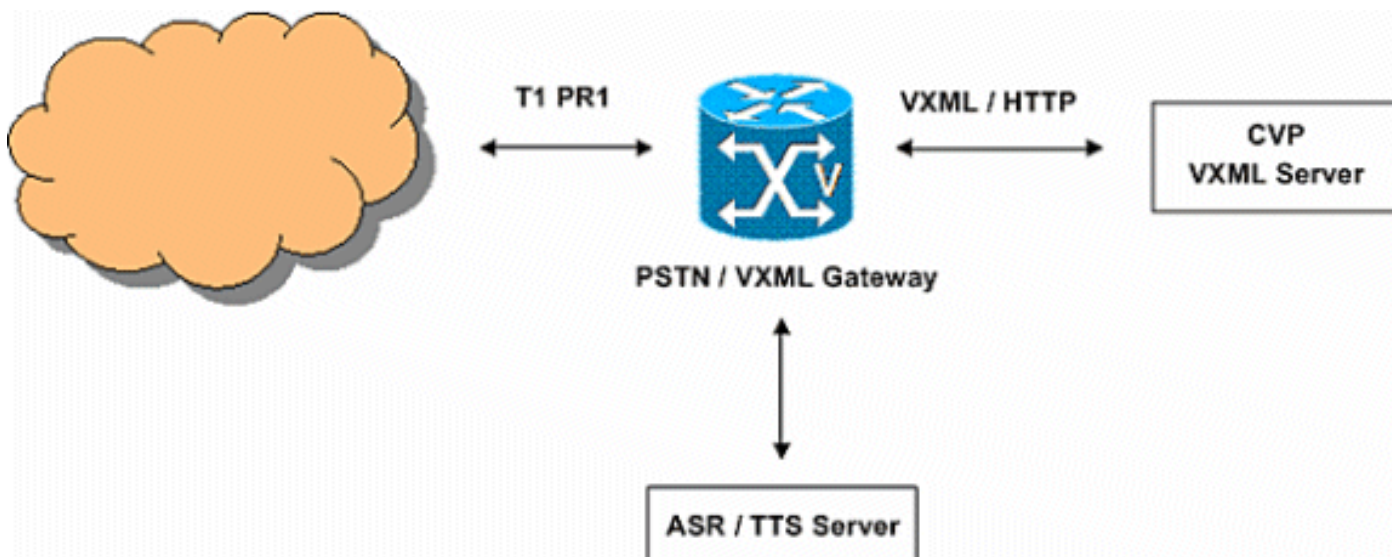
配置

本部分提供有关如何配置本文档所述功能的信息。

注意： 使用 [命令查找工具](#) ([仅限注册用户](#)) 可获取有关本部分所使用命令的详细信息。

网络图

本文档使用以下网络设置：



配置

本文档使用以下配置：

```
VXML网关配置

!---- Define Hostname to IP Address !---- mapping for ASR
```

```

and TTS servers ip host asr-en-us 172.18.110.76 ip host
tts-en-us 172.18.110.76 !--- Define the Voice class URI
to match !---- the SIP URI of ASR Server in the dial-
peer voice class uri TTS sip pattern tts@172.18.110.76
!--- Define the Voice class URI to match !---- the SIP
URI of TTS server in the dial-peer voice class uri ASR
sip pattern asr@172.18.110.76 !--- Define the amount of
maximum memory !---- to used for downloaded prompts ivr
prompt memory 15000 !--- Define the SIP URI of ASR !----
and TTS Server ivr asr-server sip:asr@172.18.110.76 ivr
tts-server sip:tts@172.18.110.76 !--- Configure an
application service for !---- CVP VXML
CVPSelfServiceBootstrap.vxml application service
CVPSelfService flash: CVPSelfServiceBootstrap.vxml
paramspace english language en paramspace english index
0 paramspace english location flash: paramspace english
prefix en !--- Configure an application service for !---
- CVP VXML CVPSelfService.tcl Script !---
CVPSelfService-app parameter specifies !---- the name of
the VXML Application !--- CVPPPrimary parameter specifies
the !---- IP address of the VXML server service Pharmacy
flash:CVPSelfService.tcl paramspace english index 0
paramspace english language en paramspace english
location flash: param CVPSelfService-port 7000 param
CVPSelfService-app GoodPrescriptionRefillApp7 paramspace
english prefix en param CVPPPrimaryVXMLServer
172.18.110.75 !--- Specifies the Gateway's RTP !----
stream to the ASR / TTS to go around the !---- Content
Service Switch !---- instead of through the CSS. mrcp
client rtpsetup enable !--- Specify the maximum memory
size !---- for the HTTP Client Cache http client cache
memory pool 15000 !--- Specify the maximum number of
file !---- that can be stored in the !---- HTTP Client
Cache http client cache memory file 500 !--- Disable
Persistent !---- HTTP Connections no http client
connection persistent !--- Configure the T1 PRI
controller T1 3/0 framing esf linecode b8zs pri-group
timeslots 1-24 !--- Configure the ISDN switch !---- type
and incoming-voice !---- under the D-channel interface
interface Serial3/0:23 no ip address encapsulation hdlc
isdn switch-type primary-net5 isdn incoming-voice modem
no cdp enable ! --- Configure a POTS !---- dial-peer
that will be used !---- as inbound dial-peer for calls
coming ! --- in across the T1 PRI line. !---- The
"pharmacy"service !---- is applied under this dial-peer.
dial-peer voice 1 pots service pharmacy destination-
pattern 5555 direct-inward-dial port 3/0:D forward-
digits all !--- Configure a SIP Voip !---- dial-peer
that will be used !---- as an outbound dial-peer when
the !---Gateway initiates a MRCP overc SIP !---- session
to the ASR server. !---- Codec = G711ulaw, DTMF-Relay !-
--- = RTP-NTE, No Vad dial-peer voice 5 voip session
protocol sipv2 destination uri ASR dtmf-relay rtp-nte
codec g711ulaw no vad !--- Configure a SIP Voip !----
dial-peer that will be used !---- as an outbound dial-
peer when the !---Gateway initiates a MRCP !---- overc
SIP session to the TTS server !--- Codec = G711ulaw,
DTMF-Relay = RTP-NTE, !---- No Vad dial-peer voice 6
voip session protocol sipv2 destination uri TTS dtmf-
relay rtp-nte codec g711ulaw no vad

```

呼叫流示例

此部分描述该的呼叫流从此配置示例的结果。

1. ISDN呼叫到达在T1PRI 3/0间的PSTN/VXML网关。
2. IOS网关匹配POTS拨号对等1作为此呼叫的呼入拨号对端。
3. IOS网关递交呼叫控制对关联给dial-peer 1.的药房服务。
4. CVP VXML/TCL写脚本关联与药房服务发送HTTP GET请求到VXML服务器。
5. VXML服务器返回200 OK答复。此答复包含VXML文档/页。
6. IOS网关执行VXML文档。
7. 如果VXML文档指定音频提示的URL，IOS网关下载音频文件并且示出提示符。
8. 如果VXML文档指定音频提示的一个文本，IOS网关建立有tts@172.18.110.76的(TTS服务器)一个SIP会话使用dial-peer 5.。在SIP会话建立后，它打开对TTS服务器的一TCP连接使用在SIP的200 OK答复SDP提供的TCP端口号邀请。此TCP连接用于交换MRCP消息例如发言，在IOS网关和TTS服务器之间的SPEAK-COMPLETE。TTS服务器发送G.711ulaw RTP音频流对IP地址，并且在SIP的SDP的网关提供的UDP端口号邀请。
9. 如果VXML文档指定网关认可DTMF位和所说的话，IOS网关建立有asr@172.18.110.76的(ASR服务器)一个SIP会话与dial-peer 6.。在SIP会话建立后，它打开对ASR服务器的一TCP连接使用在SIP的200 OK答复SDP提供的TCP端口号邀请。此TCP连接用于交换MRCP消息例如定义了语法，完成，识别和在IOS网关和ASR服务器之间的RECOGNITION-COMPLETE。IOS VXML网关发送G.711ulaw RTP音频流到在SIP 200 OK答复的SDP的ASR和UDP端口号提供的IP地址。IOS VXML网关由PSTN用户发送输入的数字作为RTP-NTE事件到ASR服务器。
10. 在VXML文档的执行，网关在VXML文档/页的<submit>标记上指定发送HTTP POST请求(与一套参数)后。
11. 步骤6 – 10为服务器寄发的每个VXML文档发生。
12. 当VXML应用程序完成为呼叫方时提供的服务，寄发与一<exit/>标记的一个VXML文档在<form>元素内。
13. IOS网关断开用TTS和ASR服务器建立的MRCPv2会话。
14. IOS网关断开在ISDN侧的呼叫。

验证

使用本部分可确认配置能否正常运行。

[命令输出解释程序 \(仅限注册用户 \)](#) (OIT) 支持某些 **show** 命令。使用 OIT 可查看对 show 命令输出的分析。

• Show call active voice brief

```
11F8 : 160 333356110ms.  
  1 +10 pid:1 Answer 5555 active  
dur 00:00:54 tx:1740/300598 rx:364/85472  
Tele 3/0:D (160) [3/0.1]  
  tx:15145/15145/0ms None noise:-52  
  acom:6  i/0:-32/-64 dBm
```

```
Telephony call-legs: 1  
SIP call-legs: 0  
H323 call-legs: 0  
Call agent controlled call-legs: 0  
SCCP call-legs: 0  
Multicast call-legs: 0  
Media call-legs: 0
```

Total call-legs: 1

• Show call active media摘要

```
11F8 : 163 333360880ms.1
  +60 pid:6 Originate
  sip:tts@172.18.110.76:5060 active
dur 00:00:44 tx:0/0 rx:2212/353545
IP 172.18.110.76:10000 SRTP:
  off rtt:0ms pl:
  4485/0ms lost:0/1/0 delay:65/65/65ms
  g711ulaw TextRelay: off
media inactive detected:n
  media contrl rcvd:
  n/a timestamp:n/a
long duration call detected:n
  long duration
  call duration:n/a timestamp:n/allF8 :
  164 333360890ms.1 +20 pid:5 Originate
  sip:asr@172.18.110.76:5060 active

dur 00:00:44 tx:1687/297152 rx:0/0
IP 172.18.110.76:10002 SRTP:
  off rtt:0ms
  pl:6550/30ms lost:0/2/0 delay:65/65/65ms
  g711ulaw TextRelay: off
media inactive detected:n media contrl
  rcvd:n/a timestamp:n/a
long duration call detected:n
  long duration
  call duration:n/a timestamp:n/a
```

```
Telephony call-legs: 0
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Media call-legs: 2
Total call-legs: 2
```

• 显示mrcc客户端会话活动详细信息

No Of Active MRCP Sessions: 1

Call-ID: 0xA0 same: 0

Resource Type: Synthesizer
URL: sip:tts@172.18.110.76
Method In Progress: SPEAK
State: S_SYNTH_SPEAKING

Associated CallID: 0xA3
MRCP version: 2.0
Control Protocol: TCP Server IP Address:
172.18.110.76 Port: 51000

Data Protocol: RTP Server IP Address:
172.18.110.76 Port: 10000
Signalling URL: sip:tts@172.18.110.76:5060

Packets Transmitted: 0 (0 bytes)
Packets Received: 2265 (361968 bytes)
ReceiveDelay: 65 LostPackets: 0

Resource Type: Recognizer
URL: sip:asr@172.18.110.76
Method In Progress: RECOGNIZE
State: S_RECOG_RECOGNIZING

Associated CallID: 0xA4
MRCP version: 2.0
Control Protocol: TCP Server IP Address:
172.18.110.76 Port: 51001

Data Protocol: RTP Server IP Address:
172.18.110.76 Port: 10002

Packets Transmitted: 1791 (313792 bytes)
Packets Received: 0 (0 bytes)
ReceiveDelay: 60 LostPackets: 0

• Show voip rtp connections

VoIP RTP active connections :
No. CallId dstCallId LocalRTP
RmtRTP LocalIP
RemoteIP
1 163 160 18964
10000 14.1.16.25
172.18.110.76
2 164 160 23072
10002 14.1.16.25
172.18.110.76
Found 2 active RTP connections

• Show http client cache

HTTP Client cached information
=====
Maximum memory pool allowed for
HTTP Client caching
= 15000 K-bytes
Maximum file size allowed for caching
= 500 K-bytes
Total memory used up for Cache
= 410 Bytes
Message response timeout = 10 secs
Total cached entries = 1
Total non-cached entries = 0

Cached entries
=====

entry 114, 1 entries
Ref FreshTime Age Size
context
--- ----- --- ----

1 86400 48 1505
0
url: http://172.18.110.75/Welcome-1.wav

本部分提供的信息可用于对配置进行故障排除。

[debug 命令](#)

配置IOS网关记录在其操作日志缓冲区的调试和禁用“logging console”。

注意： 使用 debug 命令之前，请参阅[有关 Debug 命令的重要信息](#)。

注意： 这些命令用于配置网关，来将debug存储在网关的操作日志缓冲区中：

- service timestamps debug datetime msec
- 服务顺序
- no logging console
- logging buffered 5000000 debug
- clear log

下列是用于的调试指令排除故障配置：

- debug isdn q931
- debug voip ccapi inout
- 调试voip应用程序vxml默认
- 调试voip应用程序vxml转储
- 调试ccsip消息
- debug mrcp详细信息
- 调试http客户端全部
- debug voip rtp会话nte named-event

[调试输出](#)

此部分为此示例呼叫流提供debug输出：

1. [网关收到从PSTN的一呼入呼叫。](#)
2. [网关匹配呼入拨号对端1。](#)
3. [呼叫被递交对药房服务。](#)
4. [呼叫在ISDN侧得到连接。](#)
5. [网关开始CVPSelfServiceBootstrap.vxml VoiceXML脚本的执行。](#)
6. [网关发送HTTP GET请求到VXML服务器。](#)
7. [网关收到从VXML服务器的一个200 OK消息。此答复消息主题包含VXML文档\(1\)。此VXML文档告诉网关作用媒体文件在媒体服务器查找的呼叫的Welcome-1.wav。](#)
8. [网关发送HTTP GET请求对媒体服务器下载Welcome-1.wav文件。](#)
9. [网关在HTTP消息主题中接收从媒体服务器的200 OK并且接收Welcome-1.wav的内容。](#)
10. [网关发送POST HTTP请求到服务器如对定义“提交” VXML Document\(1\)的选项。](#)
11. [网关接收200其POST HTTP请求的OK。消息主题包含VXML文档\(2\)。此VXML文档告诉网关播放“感谢您呼叫Audium药房”。注意此提示符需要由一个文本到发音的服务器综合。](#)
12. [网关发送HTTP POST请求如对VXML文档定义\(2\)的提交选项。](#)
13. [网关收到HTTP POST请求的200 OK答复。消息主题包含VXML文档\(3\)。告诉呼叫方输入1或说替换物的此VXML文档定义了菜单提示符， 2或说药剂师。提示符由一个文本到语音服务器综合。输入\(语音/DTMF\)使用一台自动语音识别器，被认可。](#)
14. [网关创建将用于DTMF语法/语音识别。一旦网关建立一个会话用ASR服务器，这些语法然后](#)

发送到ASR服务器。

15. 网关执行dial-peer查找设置—SIP会话用文本到语音服务器。呼出拨号对端6匹配。
16. 网关发送SIP邀请到TTS服务器。邀请消息的SDP包含音频流和MRCPv2应用程序的(speechsynth信道)媒体信息。
17. 网关执行dial-peer查找设置—SIP会话用自动语音识别服务器。呼出拨号对端5匹配。
18. 网关发送SIP邀请到ASR服务器。SDP包含音频流、DTMF中继和MRCPv2应用程序的(speechrecog信道)媒体信息。
19. 网关收到200 OK答复(对于SIP邀请)从ASR服务器。SIP的SDP邀请消息指定这些 : G711ulaw编码、IP地址和RTP端口号音频流的此RTP数据流方向属性：“recvonly”RTP-NTE根据DTMF中继网关(51001)将使用的TCP端口号建立一个MRCPv2会话用ASR服务器
20. 网关发送SIP ACK到ASR服务器，并且自动语音识别的SIP会话被设立在网关和ASR服务器之间。
21. 网关发送“DEFINE-GRAMMER” MRCP请求到ASR服务器。(一请求显示此处。)
22. 网关收到其DEFINE-GRAMMAR请求的200完整答复。
23. 网关收到200 OK答复(对于SIP邀请)从TTS服务器。SIP的SDP邀请消息指定这些 : G711ulaw编码、IP地址和RTP端口号音频流的此RTP数据流方向属性：“sendonly”RTP-NTE根据DTMF中继网关(51000)将使用的TCP端口号建立一个MRCPv2会话用TTS服务器
24. 网关发送SIP ACK到TTS服务器，并且文本到语音的SIP会话被设立在网关和TTS服务器之间。
25. 网关发送“认可” MRCP请求到ASR服务器开始DTMF/所说的话的识别。
26. ASR服务器发送一“进展中”答复(RECOGNIZE请求)对网关。
27. 网关完成Welcome-1.wav媒体文件下载，在缓存存储它，并且示出提示符给呼叫方。
28. 网关发送“发言” MRCP请求对TTS服务器播放“感谢你为呼叫”提示符。
29. TTS服务器发送对发言请求的一“进展中”答复。
30. 在发言“感谢你为呼叫”提示符后，TTS服务器传送“SPEAK-COMplete”信息。
31. 网关发送“发言” MRCP请求对TTS服务器播放“菜单”提示符(输入1或说Refil/输入2或说药剂师)。(debug输出没有显示。)
32. TTS服务器发送播放提示符的一个进展中，SPEAK-COMplete消息和完成。(debug输出没有显示。)
33. PSTN主叫方输入“1”选择替换物。网关发送此位作为RTP-NTE事件到ASR服务器。
34. ASR服务器传送“RECOGNITION-COMplete”信息到网关通知网关认可了其中一个请求的事件(在这种情况下位1)。
35. 在它接收从ASR服务器后的一个成功的识别通知，VXML网关在提交标记VXML文档上指定发送HTTP POST请求(3)。此POST请求通知VXML服务器位1由PSTN主叫方进入。
36. VXML服务器然后寄发要求呼叫方输入处方此处的另一个VXML文档。(debug输出没有显示。)
37. 网关传送MRCP信息对TTS发言提示符。(debug输出没有显示，但是他们类似于步骤28-30。)
38. 网关传送MRCP信息对ASR由用户检测发言的4个位处方编号。(debug输出没有显示，但是他们类似于步骤25-26。)
39. ASR认可4个位处方编号并且传送“RECOGNITION-COMplete” MRCP信息到IOS VXML网关。
40. 网关通知处方编号到VXML服务器通过发送HTTP POST请求。(debug输出没有显示，但是他们类似于步骤35。)
41. 收集运送时间和通知呼叫方的VXML服务器发送VXML页处方将是为pickup准备。网关由交互作用执行这些页用TTS和ASR服务器。(debug输出没有显示。)
42. VXML服务器寄发的最终VXML文档包含<exit \>在<form>的标记。这通知网关终止VXML会话。

43. [网关终止VXML应用程序。](#)
44. [网关断开会用ASR服务器建立的SIP会话。](#)
45. [网关断开会用TTS服务器建立的SIP会话。](#)
46. [网关断开会用ISDN侧的呼叫。](#)

[从PSTN的呼入呼叫](#)

```
*Jan 18 03:34:52.735: ISDN Se3/0:23
  Q931: RX <- SETUP pd = 8  callref = 0x005A
    Bearer Capability i = 0x8090A2
      Standard = CCITT
      Transfer Capability = Speech
      Transfer Mode = Circuit
      Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98381
      Exclusive, Channel 1
    Called Party Number i = 0x81, '5555'
      Plan:ISDN, Type:Unknown
*Jan 18 03:34:52.735: //-1/2AEE8C2A801C/
  CCAPI/cc_api_display_ie_subfields:
  cc_api_call_setup_ind_common:
  cisco-username=
  ----- ccCallInfo IE subfields -----
  cisco-ani=
  cisco-anitype=0
  cisco-aniplan=0
  cisco-anipi=0
  cisco-anisi=0
  dest=5555
  cisco-desttype=0
  cisco-destplan=1
  cisco-rdie=FFFFFFFF
  cisco-rdn=
  cisco-rdntype=-1
  cisco-rdnplan=-1
  cisco-rdnpi=-1
  cisco-rdnsi=-1
  cisco-redirectreason=-1  fwd_final_type =0
  final_redirectNumber =
  hunt_group_timeout =0
```

[呼入拨号对端1匹配](#)

```
*Jan 18 03:34:52.735:
  //-1/2AEE8C2A801C/
  CCAPI/cc_api_call_setup_ind_common:
  Interface=0x664B4BA4, Call Info(
  Calling Number=(Calling Name)=(TON=Unknown,
  NPI=Unknown, Screening=Not Screened,
  Presentation=Allowed),
  Called Number=5555(TON=Unknown, NPI=ISDN),
  Calling Translated=FALSE, Subscriber
  Type Str=RegularLine,
  FinalDestinationFlag=TRUE,
  Incoming Dial-peer=1, Progress
  Indication=NULL(0),
  Calling IE Present=FALSE,
```

```
Source Trkgrp Route Label=,  
Target Trkgrp Route Label=,  
CLID Transparent=FALSE),  
Call Id=-1
```

呼叫被递交对药房服务

```
*Jan 18 03:34:52.739:  
  //127/2AEE8C2A801C/CCAPI  
  /cc_process_call_setup_ind:  
  >>>CCAPI handed cid 127 with tag 1 to app  
  "_ManagedAppProcess_Pharmacy"  
*Jan 18 03:34:52.739:  
  //127/2AEE8C2A801C/CCAPI/ccCallSetupAck:  
  Call Id=127
```

呼叫在ISDN旁拉得到连接

```
*Jan 18 03:34:52.739:  
  ISDN Se3/0:23 Q931: TX ->  
  CONNECT pd = 8 callref =  
  0x805A  
*Jan 18 03:34:52.739:  
  //127/2AEE8C2A801C/CCAPI/ccCallHandoff:  
  Silent=FALSE, Application=0x663106C4,  
  Conference Id=0xFFFFFFFF  
*Jan 18 03:34:52.743: //127//VXML:/Open_CallHandoff:
```

网关开始CVPSelfServiceBootstrap.vxml VoiceXML脚本的执行

```
*Jan 18 03:34:52.755:  
  //127/2AEE8C2A801C/VXML:  
  /vxml_vxml_proc:  
<vxml>  
  URI(abs):flash:  
  CVPSelfServiceBootstrap.vxml  
  scheme=flash  
  path=CVPSelfServiceBootstrap.vxml  
  base=  
  URI(abs):flash:  
  CVPSelfServiceBootstrap.vxml  
  scheme=flash  
  path=CVPSelfServiceBootstrap.vxml  
  lang=none version=2.0  
<script>:  
*Jan 18 03:34:52.799: //127/2AEE8C2A801C/VXML  
  :/vxml_expr_eval:  
*Jan 18 03:34:52.863: //127/2AEE8C2A801C/VXML  
  :/vxml_jse_global_switch:  
  switch to scope(application)  
<var>: namep=handoffstring  
  expr=session.handoff_string  
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML  
  :/vxml_expr_eval:  
  expr=(var handoffstring=session.
```

```
handoff_string)
<var>: namep=application expr=getValue('APP')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var application=getValue('APP'))
<var>: namep=port expr=getValue('PORT')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var port=getValue('PORT'))
<var>: namep=callid expr=getValue('CALLID')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var callid=getValue('CALLID'))
<var>: namep=servername expr=getValue('PRIMARY')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var servername=getValue('PRIMARY'))
<var>: namep=var1 expr=getValue('var1')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var var1=getValue('var1'))
<var>: namep=var2 expr=getValue('var2')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var var2=getValue('var2'))
<var>: namep=var3 expr=getValue('var3')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var var3=getValue('var3'))
<var>: namep=var4 expr=getValue('var4')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var var4=getValue('var4'))
<var>: namep=var5 expr=getValue('var5')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var var5=getValue('var5'))
<var>: namep=status expr=getValue('status')
*Jan 18 03:34:52.867: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var status=getValue('status'))
<var>: namep=prevapp expr=getValue('prevapp')
*Jan 18 03:34:52.871: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var prevapp=getValue('prevapp'))
<var>: namep=survive expr=getValue('survive')
*Jan 18 03:34:52.871: //127/2AEE8C2A801C/VXML
  :/vxml_expr_eval:
  expr=(var survive=getValue('survive'))
<var>: namep=handoffExit
```

[网关发送HTTP GET请求到VXML服务器](#)

```
*Jan 18 03:34:52.875:
  //127//HTTPC:/httpc_write_stream:
  Client write buffer fd(3):
GET /CVP/Server?application=
  GoodPrescriptionRefillApp7&callid=
  2AEE8C2A-0AFB11D6-801C0013-
  803E8C8E&session.connection.remote.uri=555
```

```
5&session.connection.local.uri=5555 HTTP/1.1
Host: 172.18.110.75:7000
Content-Type: application/x-www-form-urlencoded
Connection: close
Accept: text/vxml, text/x-vxml, application/vxml,
        application/x-vxml, application/voicexml,
        application/x-voicexml, text/plain, tex
t/html, audio/basic, audio/wav,
        multipart/form-data,
        application/octet-stream
User-Agent: Cisco-IOS-C5400/12.4
```

[网关收到从VXML服务器的200 OK消息](#)

此答复消息主题包含VXML文档(1)。VXML文档告诉网关作用媒体文件在媒体服务器查找的呼叫的Welcome-1.wav。

```
*Jan 18 03:34:52.883: processing server
  rsp msg: msg(67CA63A8)
  URL:http://172.18.110.75:7000/CVP/
  Server?application=GoodPrescription
RefillApp7&callid=2AEE8C2A-0AFB11D6-801C0013
-803E8C8E&session.connection.
  remote.uri=5555&session.connection.local.
  uri=5555, fd(3):
*Jan 18 03:34:52.883: Request msg:
  GET /CVP/Server?application=
  GoodPrescriptionRefillApp7&callid=
  2AEE8C2A-0AFB11D6-801C0013-803E8C8
E&session.connection.remote.
  uri=5555&session
  .connection.local.uri=5555 HTTP/1.1
*Jan 18 03:34:52.883:
  Message Response Code: 200
*Jan 18 03:34:52.883:
  Message Rsp Decoded Headers:
*Jan 18 03:34:52.883:
  Date:Mon, 30 Apr 2007 16:58:39 GMT
*Jan 18 03:34:52.883:
  Content-Type:text/xml;
  charset=ISO-8859-1
*Jan 18 03:34:52.883:
  Connection:close
*Jan 18 03:34:52.883:
  Set-Cookie:JSESSIONID=
  BBCE0F948ADFDB720497F587A7997538;
  Path=/CVP

*Jan 18 03:34:52.883: headers:
*Jan 18 03:34:52.883: HTTP/1.1 200 OK
Server: Apache-Coyote/1.1
Set-Cookie: JSESSIONID=BBCE0F948ADF
DB720497F587A7997538; Path=/CVP
Content-Type: text/xml;charset=ISO-8859-1
Date: Mon, 30 Apr 2007 16:58:39 GMT
Connection: close
```

```
*Jan 18 03:34:52.883: body:
*Jan 18 03:34:52.883: <?xml version="1.0"
  encoding="UTF-8"?>
<vxml version="2.0" application=
  "/CVP/Server?audium_root=true&
  calling_into=GoodPrescriptionRefillApp7"
  xml:lang="en-us">
<form id="audium_start_form">
  <block>
    <assign name="audium_vxmlLog" expr="''" />
    <assign name="audium_element
  _start_time_millisecs"
  expr="new Date().getTime()" />
    <goto next="#start" />
  </block>
</form>
<form id="start">
  <block>
    <prompt bargein="true">
      <audio src="http://172.18.110.75/
  Welcome-1.wav" />
    </prompt>
    <assign name="audium_vxmlLog"
  expr="audium_vxmlLog
  + '|||audio_group$$$' + 'initial_audio_group'
  + '^^^'
  + application.getEla
  psedTime(audium_element_start_time_millisecs)" />
    <submit next="/CVP/Server" method="post"
  namelist=" audium_vxmlLog" />
  </block>
</form>
</vxml>
```

[网关发送HTTP GET请求对媒体服务器下载Welcome-1.wav文件](#)

```
GET /Welcome-1.wav HTTP/1.1
Host: 172.18.110.75
Content-Type:
  application/x-www-form-urlencoded
Connection: close
Accept: text/vxml,
  text/x-vxml, application/vxml,
  application/x-vxml,
  application/voicexml,
  application/x-voicexml,
  text/plain, tex
t/html, audio/basic, audio/wav,
  multipart/form-data,
  application/octet-stream
User-Agent: Cisco-IOS-C5400/12.4
```

[网关在HTTP消息主题中接收从媒体服务器的200 OK并且接收Welcome-1.wav的内容](#)

```
*Jan 18 03:34:55.647:
  //127//HTTTPC:/httpc_socket_read:
*Jan 18 03:34:55.647:
  read data from the socket 3
  : first 400 bytes of data:
HTTP/1.1 200 OK
```

Content-Length: 26450
Content-Type: audio/wav
Last-Modified:
Mon, 30 Apr 2007 15:36:51 GMT
Accept-Ranges: bytes
ETag: "e0c1445f3d8bc71:2d6"
Server: Microsoft-IIS/6.0
Date: Mon, 30 Apr 2007 16:58:42 GMT
Connection: close

RIFFJg(Unprintable char...)
0057415645666D7420120001010401
F00401F00108000666163744000176700
64617461176700FFFFFFF807
FFFFFFF80FFFFFFF80F
(other hex information not shown).

[网关发送POST HTTP请求到服务器如对定义“提交”VXML Document\(1\)的选项](#)

POST /CVP/Server HTTP/1.1
Host: 172.18.110.75:7000
Content-Length: 67
Content-Type:
application/x-www-form-urlencoded
Cookie: \$Version=0; JSESSIONID=BBCE0F948
ADFDB720497F587A7997538; \$Path=/CVP
Connection: close
Accept: text/vxml, text/x-vxml,
application/vxml,
application/x-vxml,
application/voicexml,
application/x-voicexml,
text/plain, tex
t/html, audio/basic, audio/wav,
multipart/form-data,
application/octet-stream
User-Agent: Cisco-IOS-C5400/12.4

[网关接收其POST HTTP请求的200 OK](#)

消息主题包含VXML文档(2)。VXML文档告诉网关播放“感谢您呼叫Audium药房”。注意此提示符需
要由一个文本到发音的服务器综合。

```
*Jan 18 03:34:55.651:
  processing server rsp msg:
  msg(67CA6960)URL:
  http://172.18.110.75:
  7000/CVP/Server, fd(4):
*Jan 18 03:34:55.651: Request msg:
  POST /CVP/Server HTTP/1.1
*Jan 18 03:34:55.651:
  Message Response Code: 200
*Jan 18 03:34:55.651:
  Message Rsp Decoded Headers:
*Jan 18 03:34:55.651:
  Date:Mon, 30 Apr 2007 16:58:42 GMT
*Jan 18 03:34:55.651:
  Content-Type:text/xml;
  charset=ISO-8859-1
*Jan 18 03:34:55.651: Connection:close
*Jan 18 03:34:55.651: headers:
```

```
*Jan 18 03:34:55.651: HTTP/1.1 200 OK
Server: Apache-Coyote/1.1
Content-Type: text/xml; charset=ISO-8859-1
Date: Mon, 30 Apr 2007 16:58:42 GMT
Connection: close
```

```
*Jan 18 03:34:55.655: body:
*Jan 18 03:34:55.655: <?xml version="1.0"
  encoding="UTF-8"?>
<vxml version="2.0" application=
  "/CVP/Server?audium_root=true&
  calling_into=GoodPrescriptionRefillApp7"
  xml:lang="en-us">
<form id="audium_start_form">
  <block>
    <assign name="audium_vxmlLog" expr="" />
    <assign name="audium_element
      _start_time_millisecs"
    expr="new Date().getTime()" />
    <goto next="#start" />
  </block>
</form>
<form id="start">
  <block>
    <prompt bargein="true">
    Thank you for calling Audium pharmacy.
    </prompt>
    <assign name="audium_vxmlLog" expr=
    "audium_vxmlLog + '|||audio_group$$$'
    + 'initial_audio_group'
    + '^'^ + application.getEla
    psedTime(audium_element_start_time_millisecs)" />
    <submit next="/CVP/Server" method="post"
    namelist=" audium_vxmlLog" />
  </block>
</form>
</vxml>
```

[网关发送HTTP POST请求如对VXML文档定义\(2\)的提交选项](#)

```
*Jan 18 03:34:55.667:
  //127//HTTTPC:/httpc_write_stream:
  Client write buffer fd(4):
POST /CVP/Server HTTP/1.1
Host: 172.18.110.75:7000
Content-Length: 67
Content-Type:
  application/x-www-form-urlencoded
Cookie: $Version=0; JSESSIONID=
  BBCE0F948ADFD720497F587A7997538;
  $Path=/CVP
Connection: close
Accept: text/vxml, text/x-vxml,
  application/vxml,
  application/x-vxml, application/voicexml,
  application/x-voicexml, text/plain, tex
t/html, audio/basic, audio/wav,
  multipart/form-data,
  application/octet-stream
User-Agent: Cisco-IOS-C5400/12.4
```

[网关收到HTTP POST请求的200 OK答复](#)

消息主题包含VXML文档(3)。告诉呼叫方输入1或说替换物的此VXML文档定义了菜单提示符，或者输入2或说药剂师。提示符由一个文本到语音服务器综合。输入(语音/DTMF)用一台自动语音识别器认可。

```
*Jan 18 03:34:57.499:
  processing server rsp msg:
  msg(67CA6B48)URL:
  http://172.18.110.75:7000/CVP/Server, fd(4):
*Jan 18 03:34:57.499: Request msg:
  POST /CVP/Server HTTP/1.1
*Jan 18 03:34:57.499:
  Message Response Code: 200
*Jan 18 03:34:57.499:
  Message Rsp Decoded Headers:
*Jan 18 03:34:57.499:
  Date:Mon, 30 Apr 2007 16:58:42 GMT
*Jan 18 03:34:57.499:
  Content-Type:text/xml;charset=ISO-8859-1
*Jan 18 03:34:57.499: Connection:close
*Jan 18 03:34:57.499: headers:
*Jan 18 03:34:57.499: HTTP/1.1 200 OK
Server: Apache-Coyote/1.1
Content-Type: text/xml;charset=ISO-8859-1
Date: Mon, 30 Apr 2007 16:58:42 GMT
Connection: close

*Jan 18 03:34:57.499: body:
*Jan 18 03:34:57.499: ... Buffer too large
  - truncated to (4096) len.
*Jan 18 03:34:57.499: <?xml version="1.0"
  encoding="UTF-8"?>
<vxml version="2.0" application=
  "/CVP/Server?audium_root=true&
  calling_into=GoodPrescriptionRefillApp7"
  xml:lang="en-us">
<property name="timeout" value="60s" />
<property name="confidencelevel" value="0.40" />
<form id="audium_start_form">
  <block>
    <assign name="audium_vxmlLog" expr="'" />
    <assign name="audium_element
  _start_time_millisecs"
  expr="new Date().getTime()" />
    <goto next="#start" />
  </block>
</form>
<form id="start">
  <block>
    <assign name="audium_vxmlLog"
  expr="audium_vxmlLog
  + '|||audio_group$$$' + 'initial_audio_group' + '^'^'
  + application.getElapsedTime
  (audium_element_start_time_millisecs)" />
    <goto nextitem="choice_fld" />
  </block>
<field name="choice_fld" modal="false">
  <property name="inputmodes" value="dtmf voice" />
  <prompt bargein="true">Say refills or press 1.
```

Or.


```
Say pharmacist or press 2.</prompt>
  <catch event="nomatch">
    <prompt bargein="true">Sorry.
```

I did not understand that.

Say refills or press 1.

```
Say pharmacist or press 2.</prompt>
  <assign name="audium_vxmlLog"
  expr="audium_vxmlLog
+ '|||nomatch$$$' + '1' + '^'^
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
  <assign name="audium_vxmlLog"
  expr="audium_vxmlLog
+ '|||audio_group$$$' + 'nomatch_audio_group'
+ '^'^ + application.getElapsedTime(
audium_element_start_time_millisecs)" />
</catch>
  <catch event="nomatch" count="2">
    <prompt bargein="true">
Sorry, I still did not get that.
```

If you are using a speaker phone.

Please use the phone keypad to make
your selection.

Press 1 for refills.

```
Press 2 to speak to a pharmacist.</prompt>
  <assign name="audium_vxmlLog"
  expr="audium_vxmlLog
+ '|||nomatch$$$' + '2' + '^'^
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
  <assign name="audium_vxmlLog"
  expr="audium_vxmlLog
+ '|||audio_group$$$' + 'nomatch_audio_group'
+ '^'^
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
</catch>
  <catch event="nomatch" count="3">
    <prompt bargein="true">Gee.
```

```
Looks like we are having some trouble.</prompt>
  <assign name="audium_vxmlLog"
  expr="audium_vxmlLog
+ '|||nomatch$$$' + '3' + '^'^
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
  <assign name="audium_vxmlLog"
  expr="audium_vxmlLog
+ '|||audio_group$$$' + 'nomatch_audio_group'
+ '^'^
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
  <var name="maxNoMatch" expr="'yes'" />
  <submit next="/CVP/Server" method="post"
  namelist="
audium_vxmlLog maxNoMatch" />
```

```
</catch>
<catch event="noinput">
  <prompt bargein="true">Sorry.
```

I did not hear that.

Say refills or press 1.

```
Say pharmacist or press 2.</prompt>
  <assign name="audium_vxmlLog"
    expr="audium_vxmlLog
+ '|||noinput$$$' + '1' + '^^^'
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
  <assign name="audium_vxmlLog"
    expr="audium_vxmlLog
+ '|||audio_group$$$' + 'noinput_audio_group'
+ '^^^' + application.getElapsedTime
(audium_element_start_time_millisecs)" />
</catch>
<catch event="noinput" count="2">
  <prompt bargein="true">I am sorry.
```

I still did not hear that.

If you are using a speaker phone.

Please use the phone keypad
to make your selection.

Press 1 for refills.

```
Press 2 to speak to a pharmacist.</prompt>
  <assign name="audium_vxmlLog"
    expr="audium_vxmlLog
+ '|||noinput$$$' + '2' + '^^^'
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
  <assign name="audium_vxmlLog"
    expr="audium_vxmlLog
+ '|||audio_group$$$' + 'noinput_
audio_group' + '^^^'
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
</catch>
<catch event="noinput" count="3">
  <prompt bargein="true">Gee.
```

Looks like we are having some trouble.</prompt>

```
  <assign name="audium_vxmlLog"
    expr="audium_vxmlLog
+ '|||noinput$$$' + '3' + '^^^'
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
  <assign name="audium_vxmlLog"
    expr="audium_vxmlLog
+ '|||audio_group$$$' + 'noinput_
audio_group' + '^^^'
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
  <var name="maxNoInput" expr="'yes'" />
  <submit next="/CVP/Server" method="post"
    namelist=" audium_vxmlLog maxNoInput" />
</catch>
```

```

    <option value="refills" dtmf="1">
prescription</option>
    <option value="refills">refills</option>
    <option value="refills">
prescription refills</option>
    <option value="refills">
refill my prescription</option>
    <option value="refills">
I want to refill my prescription</option>
    <option value="refills">
refills please</option>
    <option value="Pharmacist"
dtmf="2">Pharmacist</option>
    <option value="Pharmacist">
I want to speak to a pharmacist</option>
    <option value="Pharmacist">
pharmacist please</option>
    <filled>
        <assign name="audium_vxmlLog"
expr="audium_vxmlLog
+ '|||utterance$$$' + choice_fld$.
utterance + '^'^
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
        <assign name="audium_vxmlLog"
expr="audium_vxmlLog
+ '|||inputmode$$$' + choice_fld$.
inputmode + '^'^
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
        <assign name="audium_vxmlLog"
expr="audium_vxmlLog
+ '|||interpretation$$$' + choice_fld + '^'^
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
        <assign name="audium_vxmlLog"
expr="audium_vxmlLog
+ '|||confidence$$$' + choice_fld$.
confidence + '^'^
+ application.getElapsedTime
(audium_element_start_time_millisecs)" />
        <var name="confidence"
expr="choice_fld$.confidence" />
        <submit next="/CVP/Server" method="post"
namelist=" audium_vxmlLog confidence choice_fld" />
    </filled>
    </field>
</form>
</vxml>

```

[网关创建将用于DTMF语法/语音识别](#)

一旦网关建立一个会话用ASR服务器，这些语法然后发送到ASR服务器。

```

*Jan 18 03:34:57.523:
//127//AFW_:/vapp_asr_change_server:
asr_server=sip:asr@172.18.110.76
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar_id=session:option485@field.grammar

```

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
xml_lang=en-us

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
encoding_name=UTF-8

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
remoteupdate=0

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar=<?xml version="1.0" encoding="UTF-8"?>
<grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
xml:lang="en-us"
root="root"><rule id="root" scope="public">
prescription</rule></grammar>

*Jan 18 03:34:57.523: //-1//MRCP:/mrcp_get_ev:
***>Caller PC=0x61BE1F94, Count=339,
Event=0x63ACCCF0

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar_id=session:option486@field.grammar

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
encoding_name=UTF-8

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
remoteupdate=0

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar=<?xml version="1.0"
encoding="UTF-8"?>
<grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
mode="dtmf" root=
"root"><rule id="root" scope=
"public">1</rule></grammar>

*Jan 18 03:34:57.523: //-1//MRCP:
/mrcp_get_ev:
***>Caller PC=0x61BE1F94, Count=340,
Event=0x63ACCAE8

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar_id=session:option487@field.grammar

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
xml_lang=en-us

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
encoding_name=UTF-8

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
remoteupdate=0

*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar=<?xml version="1.0"
encoding="UTF-8"?>
<grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"

```
xml:lang="en-us"
root="root"><rule id="root" scope="public">
  refills</rule></grammar>
*Jan 18 03:34:57.523: //-1//MRCP
  :/mrcp_get_ev:
  ***>Caller PC=0x61BE1F94, Count=341,
  Event=0x63ACBC88
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  grammar_id=session:option488@field.grammar
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  xml_lang=en-us
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  encoding_name=UTF-8
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  remoteupdate=0
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  grammar=<?xml version="1.0" encoding="UTF-8"?>
  <grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
  xml:lang="en-us"
  root="root"><rule id="root" scope="public">
  prescription refills</rule></grammar>
*Jan 18 03:34:57.523: //-1//MRCP:/mrcp_get_ev:
  ***>Caller PC=0x61BE1F94, Count=342,
  Event=0x63ACBCB0
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  grammar_id=session:option489@field.grammar
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  xml_lang=en-us
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  encoding_name=UTF-8
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  remoteupdate=0
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  grammar=<?xml version="1.0"
encoding="UTF-8"?>
  <grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar" xml:
  lang="en-us" root="root">
  <rule id="root" scope="public">
  refill my prescription</rule><
/grammar>
*Jan 18 03:34:57.523: //-1//MRCP:/mrcp_get_ev:
  ***>Caller PC=0x61BE1F94,
  Count=343, Event=0x63ACBCD8
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
*Jan 18 03:34:57.523: //127//AFW_
  :/vapp_asr_define_grammar:
  grammar_id=session:option490@field.grammar
```

```
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
xml_lang=en-us
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
encoding_name=UTF-8
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
remoteupdate=0
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar=<?xml version="1.0" encoding="UTF-8"?>
<grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
xml:lang="en-us" root="root">
<rule id="root" scope="public">
I want to refill my prescription
</rule></grammar>
*Jan 18 03:34:57.523: //-1//MRCP:/mrcp_get_ev:
***>Caller PC=0x61BE1F94, Count=344,
Event=0x63ACBD00
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar_id=session:option491@field.grammar
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
xml_lang=en-us
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
encoding_name=UTF-8
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
remoteupdate=0
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar=<?xml version="1.0" encoding="UTF-8"?>
<grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
xml:lang="en-us"
root="root"><rule id="root" scope="public">
refills please</rule></grammar>
>
*Jan 18 03:34:57.523: //-1//MRCP:/mrcp_get_ev:
***>Caller PC=0x61BE1F94, Count=345,
Event=0x63ACBD28
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar_id=session:option492@field.grammar
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
xml_lang=en-us
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
encoding_name=UTF-8
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
remoteupdate=0
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar=<?xml version="1.0"
```

```
encoding="UTF-8"?>
<grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
xml:lang="en-us"
root="root"><rule id="root"
scope="public"> Pharmacist
</rule></grammar>
*Jan 18 03:34:57.523: //-1//MRCP:/mr_cp_get_ev:
***>Caller PC=0x61BE1F94, Count=346,
Event=0x63ACBB20
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar_id=session:option493@field.grammar
*Jan 18 03:34:57.523:
//127//AFW_:/vapp_asr_define_grammar:
encoding_name=UTF-8
*Jan 18 03:34:57.523:
//127//AFW_:/vapp_asr_define_grammar:
remoteupdate=0
*Jan 18 03:34:57.523:
//127//AFW_:/vapp_asr_define_grammar:
grammar=<?xml version="1.0"
encoding="UTF-8"?>
<grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
mode="dtmf" root="root">
<rule id="root" scope=
"public">2</rule></grammar>
*Jan 18 03:34:57.523: //-1//MRCP:/mr_cp_get_ev:
***>Caller PC=0x61BE1F94,
Count=347, Event=0x63ACBD50
*Jan 18 03:34:57.523:
//127//AFW_:/vapp_asr_define_grammar:
*Jan 18 03:34:57.523:
//127//AFW_:/vapp_asr_define_grammar:
grammar_id=session:
option494@field.grammar
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
xml_lang=en-us
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
encoding_name=UTF-8
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
remoteupdate=0
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
grammar=<?xml version="1.0"
encoding="UTF-8"?>
<grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
xml:lang="en-us"
root="root"><rule id="root" scope="public">
I want to speak to a pharmacist
</rule></grammar>
*Jan 18 03:34:57.523: //-1//MRCP:/mr_cp_get_ev:
***>Caller PC=0x61BE1F94,
Count=348, Event=0x63ACBFF8
*Jan 18 03:34:57.523: //127//AFW_
:/vapp_asr_define_grammar:
*Jan 18 03:34:57.527: //127//AFW_
```

```
:/vapp_asr_define_grammar:
grammar_id=session:option495@field.grammar
*Jan 18 03:34:57.527: //127//AFW_
:/vapp_asr_define_grammar:
xml_lang=en-us
*Jan 18 03:34:57.527: //127//AFW_
:/vapp_asr_define_grammar:
encoding_name=UTF-8
*Jan 18 03:34:57.527: //127//AFW_
:/vapp_asr_define_grammar:
remoteupdate=0
*Jan 18 03:34:57.527: //127//AFW_
:/vapp_asr_define_grammar:
grammar=<?xml version="1.0"
encoding="UTF-8"?>
<grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
xml:lang="en-us"
root="root"><rule id="root" scope="public">
pharmacist please
</rule></grammar>

*Jan 18 03:34:57.527:
//-1//MRCP:/mrcp_get_ev:

***>Caller PC=0x61BE1F94,
Count=349, Event=0x63ACC048
*Jan 18 03:34:57.527: //127//AFW_
:/vapp_asr_define_grammar:
*Jan 18 03:34:57.527:
//127//AFW_:/vapp_asr_define_grammar:
grammar_id=session:link496@document.grammar
*Jan 18 03:34:57.527:
//127//AFW_:/vapp_asr_define_grammar:
xml_lang=en-us
*Jan 18 03:34:57.527:
//127//AFW_:/vapp_asr_define_grammar:
encoding_name=UTF-8
*Jan 18 03:34:57.527:
//127//AFW_:/vapp_asr_define_grammar:
remoteupdate=0
*Jan 18 03:34:57.527:
//127//AFW_:/vapp_asr_define_grammar:
grammar=<?xml version="1.0"
encoding="UTF-8"?>
<grammar xmlns="http://ww
w.w3.org/2001/06/grammar" mode="voice"
version="1.0"
root="Hotlink_02_VOICE" xml:lang="en-us">
<rule id="Hotlink_02_VOICE" scope="public">
<one-of>
<item>operator</item>
<item>agent</item>
<item>pharmacist</item>
</one-of>
</rule>
</grammar>
*Jan 18 03:34:57.527: //-1//MRCP:/mrcp_get_ev:
***>Caller PC=0x61BE1F94, Count=350,
Event=0x63ACC098
*Jan 18 03:34:57.527:
//127//AFW_:/vapp_asr_define_grammar:
*Jan 18 03:34:57.527:
//127//AFW_:/vapp_asr_define_grammar:
```



```
grammar_id=session:link497@document.grammar
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
  xml_lang=en-us
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
  encoding_name=UTF-8
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
  remoteupdate=0
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
  grammar=<?xml version="1.0" encoding="UTF-8"?>
  <grammar xmlns="http://ww
w.w3.org/2001/06/grammar" mode="voice" version="1.0"
  root="Hotlink_01_VOICE" xml:lang="en-us">
    <rule id="Hotlink_01_VOICE" scope="public">
      <one-of>
        <item>operator</item>
        <item>agent</item>
        <item>pharmacist</item>
      </one-of>
    </rule>
  </grammar>
*Jan 18 03:34:57.527:
  //-1//MRCP:/mrcp_get_ev:
  ***>Caller PC=0x61BE1F94, Count=351,
  Event=0x63ACC0C0
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
  grammar_id=session:help@grammar
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
  xml_lang=en-us
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
  encoding_name=UTF-8
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
  remoteupdate=1
*Jan 18 03:34:57.527:
  //127//AFW_:/vapp_asr_define_grammar:
  grammar=<?xml version="1.0"
  encoding="UTF-8"?>
  <grammar version="1.0" xm
lns="http://www.w3.org/2001/06/grammar"
  xml:lang="en-us"
  root="root"><rule id="root"
  scope="public">
  help</rule></grammar>
*Jan 18 03:34:57.527:
  //-1//MRCP:/mrcp_get_ev:
  ***>Caller PC=0x61BE1F94, Count=352,
  Event=0x63ACBEE0
*Jan 18 03:34:57.527: //127//AFW_:/vapp_asr:
  grammar_id=session:option485@field.grammar
grammar_id=session:option486@field.grammar
grammar_id=session:option487@field.grammar
grammar_id=session:option488@field.grammar
grammar_id=session:option489@field.grammar
grammar_id=session:option490@field.grammar
grammar_id=session:option491@field.grammar
```

```
grammar_id=session:option492@field.grammar
grammar_id=session:option493@field.grammar
grammar_id=session:option494@field.grammar
grammar_id=session:option495@field.grammar
grammar_id=session:link496@document.grammar
grammar_id=session:link497@document.grammar
grammar_id=session:help@grammar
```

网关执行Dial-peer查找设置SIP会话用文本到语音服务器

呼出拨号对端6匹配。

```
*Jan 18 03:34:57.527:
  //-1/xxxxxxxxxxxx/CCAPI/ccCallSetupRequest:

  Destination Pattern=,
  Called Number=sip:tts@172.18.110.76,
  Digit Strip=FALSE

*Jan 18 03:34:57.527:
  //-1/xxxxxxxxxxxx/CCAPI/ccCallSetupRequest:

  Calling Number=5555(TON=Unknown, NPI=Unknown,
  Screening=Not Screened,

  Presentation=Allowed),

  Called Number=sip:tts@172.18.110.76(TON=Unknown,
  NPI=ISDN),

  Redirect Number=, Display Info=

  Account Number=, Final Destination Flag=TRUE,

  Guid=2AEE8C2A-0AFB-11D6-801C-0013803E8C8E,
  Outgoing Dial-peer=6

*Jan 18 03:34:57.531:
  //-1/xxxxxxxxxxxx/CCAPI/cc
  _api_display_ie_subfields:

  ccCallSetupRequest:

  cisco-username=

  ----- ccCallInfo IE subfields -----

  cisco-ani=5555

  cisco-anitype=0

  cisco-aniplan=0

  cisco-anipi=0

  cisco-anisi=0

  dest=sip:tts@172.18.110.76

  cisco-desttype=0

  cisco-destplan=1
```

```
cisco-rdie=FFFFFFFF
cisco-rdn=
cisco-rdntype=-1
cisco-rdnplan=-1
cisco-rdnpi=-1
cisco-rdnsi=-1
cisco-redirectreason=-1 fwd_final_type =0
final_redirectNumber =
hunt_group_timeout =0
```

```
*Jan 18 03:34:57.531:
  //-1/xxxxxxxxxxxx/CCAPI/
  ccIFCallSetupRequestPrivate:

  Interface=0x662CE538, Interface Type=3,
  Destination=, Mode=0x0,

  Call Params(Calling Number=5555,
  (Calling Name=)(TON=Unknown,
  NPI=Unknown, Screening=Not Screened,
  Presentation=Allowed),

  Called Number=sip:tts@172.18.110.76
  (TON=Unknown, NPI=ISDN),
  Calling Translated=FALSE,

  Subscriber Type Str=RegularLine,
  FinalDestinationFlag=TRUE,
  Outgoing Dial-peer=6, Call Count On=FALSE,

  Source Trkgrp Route Label=,
  Target Trkgrp Route Label=,
  tg_label_flag=0, Application Call Id=)
```

[网关发送SIP邀请对TTS服务器](#)

邀请消息的SDP包含音频流和MRCPv2应用程序的(speechsynth信道)媒体信息。

```
*Jan 18 03:34:57.531:
  //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:

  Sent:

  INVITE sip:tts@172.18.110.76:5060 SIP/2.0
```

Via: SIP/2.0/UDP 14.1.16.25:
5060;branch=z9hG4bK931F1D

Remote-Party-ID: <sip:5555@14.1.16.25>;
party=calling;screen=no;privacy=off

From: <sip:5555@14.1.16.25>
;tag=E54D43C-1EC4

To: sip:tts@172.18.110.76

Date: Fri, 18 Jan 2002 03:34:57 GMT

Call-ID: 2DCA5BEF-AFB11D6-80D3DC30
-3585E95A@14.1.16.25

Supported: 100rel,timer,
resource-priority,replaces

Min-SE: 1800

Cisco-Guid: 720276522-184226262
-2149318675-2151582862

User-Agent: Cisco-SIPGateway/IOS-12.x

Allow: INVITE, OPTIONS, BYE,
CANCEL, ACK, PRACK, UPDATE,
REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER

CSeq: 101 INVITE

Max-Forwards: 70

Timestamp: 1011324897

Contact: <sip:5555@14.1.16.25:5060>

Expires: 180

Allow-Events: telephone-event

Content-Type: application/sdp

Content-Disposition:
session;handling=required

Content-Length: 358

v=0

o=CiscoSystemsSIP-GW-UserAgent
6021 4611 IN IP4 14.1.16.25

s=SIP Call

c=IN IP4 14.1.16.25

t=0 0

m=audio 16984 RTP/AVP 0 101

```
c=IN IP4 14.1.16.25
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=recvonly
a=mid:1
m=application 9 TCP/MRCPv2
a=setup:active
a=connection:new
a=resource:speechsynth
a=cmid:1
```

[网关执行Dial-peer查找设置SIP会话用ASR服务器](#)

呼出拨号对端5匹配。

```
*Jan 18 03:34:57.531:
  //-1/xxxxxxxxxxxx/CCAPI/ccCallSetupRequest:

  Destination Pattern=,
  Called Number=sip:asr@172.18.110.76,
  Digit Strip=FALSE

*Jan 18 03:34:57.531:
  //-1/xxxxxxxxxxxx/CCAPI/ccCallSetupRequest:

  Calling Number=5555(TON=Unknown, NPI=Unknown,
  Screening=Not Screened, Presentation=Allowed),

  Called Number=sip:asr@172.18.110.76
  (TON=Unknown, NPI=ISDN),

  Redirect Number=, Display Info=

  Account Number=, Final Destination Flag=TRUE,

  Guid=2AEE8C2A-0AFB-11D6-801C-0013803E8C8E,
  Outgoing Dial-peer=5

*Jan 18 03:34:57.531:
  //-1/xxxxxxxxxxxx/CCAPI/cc_api
  _display_ie_subfields:

  ccCallSetupRequest:

  cisco-username=

  ----- ccCallInfo IE subfields -----
```

cisco-ani=5555
cisco-anitype=0
cisco-aniplan=0
cisco-anipi=0
cisco-anisi=0
dest=sip:asr@172.18.110.76
cisco-desttype=0
cisco-destplan=1
cisco-rdie=FFFFFFFF
cisco-rdn=
cisco-rdntype=-1
cisco-rdnplan=-1
cisco-rdnpi=-1
cisco-rdnsi=-1
cisco-redirectreason=-1
fwd_final_type =0
final_redirectNumber =
hunt_group_timeout =0

*Jan 18 03:34:57.535:

//-1/xxxxxxxxxxxxx/CCAPI
/ccIFCallSetupRequestPrivate:

Interface=0x662CE538, Interface Type=3,
Destination=, Mode=0x0,

Call Params(Calling Number=5555,
(Calling Name=)(TON=Unknown,
NPI=Unknown, Screening=Not Screened,
Presentation=Allowed),

Called Number=sip:asr@172.18.110.76
(TON=Unknown, NPI=ISDN),
Calling Translated=FALSE,

Subscriber Type Str=RegularLine,
FinalDestinationFlag=TRUE,
Outgoing Dial-peer=5, Call Count On=FALSE,

Source Trkgrp Route Label=,
Target Trkgrp Route Label=,
tg_label_flag=0, Application Call Id=)

[网关发送SIP邀请到ASR服务器](#)

SDP包含音频流的媒体信息，DTMF中继。并且MRCPv2应用程序(speechrecog信道)。

```
*Jan 18 03:34:57.535:
  //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
```

Sent:

```
INVITE sip:asr@172.18.110.76:5060 SIP/2.0
```

```
Via: SIP/2.0/UDP
  14.1.16.25:5060;branch=z9hG4bK94C0B
```

```
Remote-Party-ID: <sip:5555@14.1.16.25>;
  party=calling;screen=no;privacy=off
```

```
From: <sip:5555@14.1.16.25>;tag=E54D440-1CDB
```

```
To: sip:asr@172.18.110.76
```

```
Date: Fri, 18 Jan 2002 03:34:57 GMT
```

```
Call-ID: 2DCAF817-AFB11D6
  -80D5DC30-3585E95A@14.1.16.25
```

```
Supported: 100rel,timer,
  resource-priority,replaces
```

```
Min-SE: 1800
```

```
Cisco-Guid: 720276522-184226262-
  2149318675-2151582862
```

```
User-Agent: Cisco-SIPGateway/IOS-12.x
```

```
Allow: INVITE, OPTIONS, BYE, CANCEL,
  ACK, PRACK, UPDATE,
  REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
```

```
CSeq: 101 INVITE
```

```
Max-Forwards: 70
```

```
Timestamp: 1011324897
```

```
Contact: <sip:5555@14.1.16.25:5060>
```

```
Expires: 180
```

```
Allow-Events: telephone-event
```

```
Content-Type: application/sdp
```

```
Content-Disposition:
  session;handling=required
```

```
Content-Length: 358
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent
 6805 2057 IN IP4 14.1.16.25
s=SIP Call
c=IN IP4 14.1.16.25
t=0 0
m=audio 19994 RTP/AVP 0 101
c=IN IP4 14.1.16.25
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=sendonly
a=mid:1
m=application 9 TCP/MRCPv2
a=setup:active
a=connection:new
a=resource:speechrecog
a=cmid:1
```

网关收到200 OK答复(对于SIP请邀请)从ASR服务器

1. G711ulaw编码、IP地址和RTP端口号音频流的。
2. 此RTP数据流方向属性是“recvonly”。
3. RTP-NTE根据DTMF中继。
4. 网关(51001)将使用的TCP端口号建立一个MRCPv2会话用ASR服务器。

```
*Jan 18 03:34:57.559:
  //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
```

Received:

SIP/2.0 200 OK

Via: SIP/2.0/UDP 14.1.16.25:5060;
branch=z9hG4bK94C0B

To: <sip:asr@172.18.110.76>;tag=a99d0500

From: <sip:5555@14.1.16.25>;tag=E54D440-1CDB

Call-ID: 2DCAF817-AFB11D6-80D5DC30-

3585E95A@14.1.16.25

CSeq: 101 INVITE

Contact: <sip:172.18.110.76:5060>

Content-Type: application/sdp

Content-Length: 342

v=0

o=MRCpv2Server 3386937590 3386937590
IN IP4 172.18.110.76

s=SIP Call

c=IN IP4 172.18.110.76

t=3386937590 0

m=audio 10002 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=recvonly

m=application 51001 TCP/MRCpv2

a=connection:new

a=setup:passive

a=model:besteffort

a=channel:000023B846361276@speechrecog

网关发送SIP ACK到ASR服务器

ASR的SIP会话被设立在网关和ASR服务器之间。

*Jan 18 03:34:57.563:

//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:

Sent:

ACK sip:172.18.110.76:5060 SIP/2.0

Via: SIP/2.0/UDP 14.1.16.25:5060;branch=z9hG4bK9520FA

From: <sip:5555@14.1.16.25>;tag=E54D440-1CDB

To: <sip:asr@172.18.110.76>;tag=a99d0500

Date: Fri, 18 Jan 2002 03:34:57 GMT

Call-ID: 2DCAF817-AFB11D6-80D5DC30-3585E95A@14.1.16.25

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: telephone-event

Content-Length: 0

[网关发送“DEFINE-GRAMMER” MRCP请求到ASR服务器](#)

—请求显示此处。

MRCP/2.0 446 DEFINE-GRAMMAR 1

Channel-Identifier: 000023B846361276@speechrecog

:

Speech-Language: en-us

Content-Base: http://172.18.110.75:7000/CVP/

:

Content-Type: application/srgs+xml

Content-Id: option485@field.grammar

Content-Length: 193

:

```
<?xml version="1.0" encoding="UTF-8"?>
  <grammar version="1.0"
    xmlns="http://www.w3.org/2001/06/grammar"
    xml:lang="en-us" root="root">
    <rule id="root" scope="public">
      prescription</rule></grammar>
```

[网关收到其DEFINE-GRAMMAR请求的200完整答复](#)

*Jan 18 03:34:57.587: //-1//MRCP:/hash_get:

Table=mrpcv2_socket_connect_table, Key=0:

MRCP/2.0 80 1 200 COMPLETE

Channel-Identifier: 000023B846361276@speechrecog

[网关收到200 OK答复\(对于SIP请邀请\)从TTS服务器](#)

SIP的SDP邀请消息指定这些：

1. G711ulaw编码、IP地址和RTP端口号音频流的。
2. 此RTP数据流方向属性是“sendonly”。
3. RTP-NTE根据DTMF中继
4. 网关(51000)将使用的TCP端口号建立一个MRCPv2会话用TTS服务器。

```
*Jan 18 03:34:57.591:
  //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
```

Received:

SIP/2.0 200 OK

Via: SIP/2.0/UDP 14.1.16.25:5060;
branch=z9hG4bK931F1D

To: <sip:tts@172.18.110.76>;tag=c1160600

From: <sip:5555@14.1.16.25>;tag=E54D43C-1EC4

Call-ID: 2DCA5BEF-AFB11D6-80D3DC30-
3585E95A@14.1.16.25

CSeq: 101 INVITE

Contact: <sip:172.18.110.76:5060>

Content-Type: application/sdp

Content-Length: 342

v=0

o=MRCPv2Server 3386937590 3386937590
IN IP4 172.18.110.76

s=SIP Call

c=IN IP4 172.18.110.76

t=3386937590 0

m=audio 10000 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=sendonly

m=application 51000 TCP/MRCPv2

a=connection:new

a=setup:passive

a=model:besteffort

a=channel:000023EC46361276@speechsynth

网关发送SIP ACK到TTS服务器

文本到语音的SIP会话被设立在网关和TTS服务器之间。

```
*Jan 18 03:34:57.595:
  //-1/xxxxxxxxxxxx/SIP/
  Msg/ccsipDisplayMsg:

Sent:

ACK sip:172.18.110.76:5060 SIP/2.0

Via: SIP/2.0/UDP 14.1.16.25:5060;
     branch=z9hG4bK9626BC

From: <sip:5555@14.1.16.25>;tag=E54D43C-1EC4

To: <sip:tts@172.18.110.76>;tag=c1160600

Date: Fri, 18 Jan 2002 03:34:57 GMT

Call-ID: 2DCA5BEF-AFB11D6-80D3DC30
        -3585E95A@14.1.16.25

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: telephone-event

Content-Length: 0
```

网关发送“认可” MRCP请求到ASR服务器

```
MRCP/2.0 987
  RECOGNIZE 15

Channel-Identifier:
  000023B846361276@speechrecog

:

Speech-Language: en-us

Confidence-Threshold: 0.40

Sensitivity-Level: 0.50

Speed-Vs-Accuracy: 0.50

Cancel-If-Queue: false

Dtmf-Interdigit-Timeout: 10000

Dtmf-Term-Timeout: 0
```

Dtmf-Term-Char: #
No-Input-Timeout: 60000
N-Best-List-Length: 1
Logging-Tag: 127:127
Accept-Charset: charset: utf-8
Content-Base:
 http://172.18.110.75:7000/CVP/
Media-Type: audio/basic
Start-Input-Timers: false
:
Content-Type: text/uri-list
Content-Length: 453

:
session:option485@field.grammar
session:option486@field.grammar
session:option487@field.grammar
session:option488@field.grammar
session:option489@field.grammar
session:option490@field.grammar
session:option491@field.grammar
session:option492@field.grammar
session:option493@field.grammar
session:option494@field.grammar
session:option495@field.grammar
session:link496@document.grammar
session:link497@document.grammar
session:help@grammar

[ASR服务器发送“进展中”答复\(为请认可请求\)对网关](#)

MRCP/2.0 84 15 200 IN-PROGRESS

Channel-Identifier:
000023B846361276@speechrecog

网关完成Welcome-1.wav媒体文件的下载

它在缓存存储它并且示出提示符给呼叫方。

```
*Jan 18 03:35:04.335:
  //127//HTTPC:/httpc_is_cached:
  HTTPC_FILE_IS_CACHED

*Jan 18 03:35:04.335: //-1//HTTPC:
  /httpc_set_cache_revoke_cb:
  Registering revoke_callback(0x61CDD948)
  +pcontext(0x63A7AAA8) for cach

ep(0x68734930)

*Jan 18 03:35:04.335: //127//AFW_:/vapp_driver:
  evtID: 146 vapp record state: 0

*Jan 18 03:35:04.335: //127//AFW_:/vapp_play_done:
  evID=146 reason=17,
  protocol=5, status_code=0, dur=3291, rate=0

*Jan 18 03:35:04.335: //127/2AEE8C2A801C/VXML:
  /vxml_media_done:
```

网关发送“发言” MRCP请求对TTS服务器播放感谢提示符

```
MRCP/2.0 376      SPEAK 1

Channel-Identifier:
  000023EC46361276@speechsynth

:

Kill-On-Barge-In: true

Speech-Language: en-us

Logging-Tag: 127:127

Content-Base:
  http://172.18.110.75:7000/CVP/

:

Content-Type: application/ssml+xml

Content-Length: 123

:
```

```
<?xml version="1.0" encoding="UTF-8"?>
  <speak version="1.0" xml:lang="en-us">
    Thank you for calling Audium pharmacy.</speak>
```

TTS服务器发送发言请求的“进展中”答复

MRCP/2.0 83 1 200 IN-PROGRESS

Channel-Identifier:
000023EC46361276@speechsynth

在发言感谢提示符后，TTS服务器传送“SPEAK-COMplete”信息

MRCP/2.0 141 SPEAK-COMplete 1 COMplete

Channel-Identifier:
000023EC46361276@speechsynth

Completion-Cause: 000 normal

Speech-Marker: ""

PSTN主叫方输入“1”选择替换物

网关发送此位作为RTP-NTE事件到ASR服务器。

```
*Jan 18 03:35:12.583:
  s=DSP d=VoIP payload 0x65 ssrc
  0x15 sequence 0x1E9B timestamp 0x2FADCC60

*Jan 18 03:35:12.583:          Pt:101    Evt:1
  Pkt:03 00 00 <Snd>>>

*Jan 18 03:35:12.587:
  s=DSP d=VoIP payload 0x65 ssrc
  0x15 sequence 0x1E9C timestamp 0x2FADCC60

*Jan 18 03:35:12.587:          Pt:101    Evt:1
  Pkt:03 00 00 <Snd>>>

*Jan 18 03:35:12.631:
  s=DSP d=VoIP payload 0x65 ssrc
  0x15 sequence 0x1E9E timestamp 0x2FADCC60

*Jan 18 03:35:12.631:          Pt:101    Evt:1
  Pkt:03 01 90 <Snd>>>

*Jan 18 03:35:12.683:
  s=DSP d=VoIP payload 0x65 ssrc
  0x15 sequence 0x1E9F timestamp 0x2FADCC60

*Jan 18 03:35:12.683:          Pt:101    Evt:1
  Pkt:03 03 20 <Snd>>>

*Jan 18 03:35:12.703:
```

s=DSP d=VoIP payload 0x65 ssrc
0x15 sequence 0x1EA0 timestamp 0x2FADCC60

*Jan 18 03:35:12.703: Pt:101 Evt:1
Pkt:83 03 38 <Snd>>>

*Jan 18 03:35:12.707: s=DSP d=VoIP payload
0x65 ssrc 0x15 sequence 0x1EA1 timestamp 0x2FADCC60

*Jan 18 03:35:12.707: Pt:101 Evt:1
Pkt:83 03 38 <Snd>>>

*Jan 18 03:35:12.711: s=DSP d=VoIP payload
0x65 ssrc 0x15 sequence
0x1EA2 timestamp 0x2FADCC60

*Jan 18 03:35:12.711: Pt:101 Evt:1
Pkt:83 03 38 <Snd>>>

[ASR服务器传送“RECOGNITION-COMplete”信息到网关](#)

这通知网关认可了其中一个请求的事件(在这种情况下位1)。

MRCP/2.0 513

RECOGNITION-COMplete 15 COMplete

Channel-Identifier:

000023B846361276@speechrecog

Proxy-Sync-Id: 0B82553000000027

Completion-Cause: 000 success

Content-Type: application/nlsml+xml

Content-Length: 292

<?xml version="1.0" encoding="UTF-8"?>

<result grammar="session:option486@field.grammar">

<interpretation grammar=
"session:option486@field.grammar"
confidence="0.000000">

<instance>

1

</instance>

<input mode="dtmf"
confidence="1.000000">

1

</input>

</interpretation>

</result>

VXML网关接收从ASR服务器的成功的识别通知

在此通知收据，VXML网关在提交标记VXML文档上指定发送HTTP POST请求(3)后。此POST请求通知VXML服务器位1由PSTN主叫方进入。

*Jan 18 03:35:12.863:

//127/2AEE8C2A801C/VXML:/vxml_vapp_bgpost:

url http://172.18.110.75:7000/CVP/Server

cachable 1 timeout

0 body audium_vxmlLog=%7C%7C%7Caudio

_group\$\$\$initial_audio_group%5E%

5E%5E4%7C%7C%7Cutterance\$\$\$1%5E%5E%5E153

40%7C%7C%7Cinputmode

\$\$dtmf%5E%5E%5E15344%7C%7C%7C

interpretation\$\$\$refills%5E%5E%5E15344%7C

%7C%7Cconfidence\$\$\$0%5E%5E%5E15344&confidence=

0&choice_fld=refills

len 258maxage -1 maxstale -1

*Jan 18 03:35:12.863: //127//AFW_:/vapp_bgpost:

url=http://172.18.110.75:7000/CVP/Server;

mime_type=application/x-www-form-urlencoded

ed; len=258; iov_base=audium_vxmlLog=%7C%7C%7Caudio_

group\$\$\$initial_audio_group

%5E%5E%5E4%7C%7C%7Cutterance

\$\$\$1%5E%5E%5E15340%7C%7C

%7Cinputmode\$\$dtmf%5E%5E%5E15344%

7C%7C%7Cinterpretation\$\$\$refills

%5E%5E%5E15344%7C%7C%7Cconfidence\$\$\$0

%5E%5E%5E15344&confidence=0&

choice_fld=refills

*Jan 18 03:35:12.931:

about to send data to the socket 3

: first 400 bytes of data:

POST /CVP/Server HTTP/1.1

Host: 172.18.110.75:7000

Content-Length: 258

Content-Type: application/x-www-form-urlencoded

Cookie: \$Version=0; JSESSIONID=

BBCE0F948ADFDB720497F587A7997538;

\$Path=/CVP

Connection: close

Accept: text/vxml, text/x-vxml, application/vxml,
application/x-vxml,
application/voicexml, application/x-voicexml,
text/plain, tex

t/html, audio/basic, audio/wav, multipart/form-dat

ASR认可四位数字的处方编号

ASR传送RECOGNITION-COMPLETE MRCP信息到IOS VXML网关。

MRCP/2.0 533

RECOGNITION-COMPLETE 21 COMPLETE

Channel-Identifier:

000023B846361276@speechrecog

Proxy-Sync-Id: 0B82553000000028

Completion-Cause: 000 success

Content-Type: application/nlsml+xml

Content-Length: 312

<?xml version="1.0" encoding="UTF-8"?>

<result grammar=

"session:field498@field.grammar">

<interpretation grammar=

"session:field498@field.grammar"

confidence="0.738968">

<instance>

1234

</instance>

<input mode="speech"

confidence="0.752155">

one two three four

</input>

</interpretation>

</result>

The final VXML document sent by the
VXML server contains just the

<exit\> tag in the <form>

This tells the Gateway to
terminate the VXML session

VXML服务器寄发的最后VXML文档包含退出标记以形式

这通知网关终止VXML会话

```
*Jan 18 03:36:07.159:
  processing server rsp msg:
  msg(67CA85F8)URL:
  http://172.18.110.75:7000/CVP/Server, fd(3):
```

```
*Jan 18 03:36:07.159: Request msg:
  POST /CVP/Server HTTP/1.1
```

```
*Jan 18 03:36:07.159:
  Message Response Code: 200
```

```
*Jan 18 03:36:07.159:
  Message Rsp Decoded Headers:
```

```
*Jan 18 03:36:07.159: D
  ate:Mon, 30 Apr 2007 16:59:53 GMT
```

```
*Jan 18 03:36:07.159:
  Content-Type:text/xml;charset=ISO-8859-1
```

```
*Jan 18 03:36:07.159: Connection:close
```

```
*Jan 18 03:36:07.159: Set-Cookie:
  JSESSIONID=NULL;
  Expires=Thu, 01-Jan-1970
  00:00:10 GMT; Path=/CVP
```

```
*Jan 18 03:36:07.159: headers:
```

```
*Jan 18 03:36:07.159: HTTP/1.1 200 OK
```

```
Server: Apache-Coyote/1.1
```

```
Set-Cookie: JSESSIONID=NULL; Expires=Thu,
  01-Jan-1970 00:00:10 GMT; Path=/CVP
```

```
Content-Type: text/xml;charset=ISO-8859-1
```

```
Date: Mon, 30 Apr 2007 16:59:53 GMT
```

```
Connection: close
```

```
*Jan 18 03:36:07.159: body:
```

```
*Jan 18 03:36:07.159: <?xml version="1.0"
  encoding="UTF-8"?>
```

```
<vxml version="2.0" xml:lang="en-us">

  <catch event="vxml.session.error">

    <exit />

  </catch>

  <catch event="telephone.disconnect.hangup">

    <exit />

  </catch>

  <catch event="telephone.disconnect">

    <exit />

  </catch>

  <catch event="error.unsupported.object">

    <exit />

  </catch>

  <catch event="error.unsupported.language">

    <exit />

  </catch>

  <catch event="error.unsupported.format">

    <exit />

  </catch>

  <catch event="error.unsupported.element">

    <exit />

  </catch>

  <catch event="error.unsupported.builtin">

    <exit />

  </catch>

  <catch event="error.unsupported">

    <exit />

  </catch>

  <catch event="error.semantic">

    <exit />

  </catch>

  <catch event="error.noresource">
```

```
<exit />

</catch>

<catch event="error.noauthorization">

  <exit />

</catch>

<catch event="error.eventhandler.notfound">

  <exit />

</catch>

<catch event="error.connection.noroute">

  <exit />

</catch>

<catch event="error.connection.noresource">

  <exit />

</catch>

<catch event="error.connection.nolicense">

  <exit />

</catch>

<catch event="error.connection.noauthorization">

  <exit />

</catch>

<catch event="error.connection.baddestination">

  <exit />

</catch>

<catch event="error.condition.baddestination">

  <exit />

</catch>

<catch event="error.com.cisco.
media.resource.unavailable">

  <exit />

</catch>

<catch event=
"error.com.cisco.handoff.failure">

  <exit />
```

```
</catch>

<catch event=
  "error.com.cisco.callhandoff.failure">

  <exit />
</catch>

<catch event=
  "error.com.cisco.aaa.authorize.failure">

  <exit />
</catch>

<catch event=
  "error.com.cisco.aaa.authenticate.failure">

  <exit />
</catch>

<catch event="error.badfetch.https">

  <exit />
</catch>

<catch event="error.badfetch.http">

  <exit />
</catch>

<catch event="error.badfetch">

  <exit />
</catch>

<catch event="error">

  <exit />
</catch>

<catch event="disconnect.com.cisco.handoff">

  <exit />
</catch>

<catch event="connection.disconnect.hangup">

  <exit />
</catch>

<catch event="connection.disconnect">

  <exit />
</catch>
```

```
<form>
  <block>
    <exit />
  </block>
</form>
</vxml>
```

网关终止VXML应用程序

```
*Jan 18 03:36:14.155:
  //127/2AEE8C2A801C/VXML:/vxml_vapp_terminate:

  vapp_status=0 ref_count 0

*Jan 18 03:36:14.155:
  //127//AFW_:/vapp_terminate:

*Jan 18 03:36:14.155: //127//AFW_
  :/vapp_session_exit_event_name:
  Exit Event vxml.session.complete

*Jan 18 03:36:14.155:
  //127//AFW_:/AFW_M_VxmlModule_Terminate:

*Jan 18 03:36:14.155:
  //131/2AEE8C2A801C/CCAPI/ccCallDisconnect:

  Cause Value=16, Tag=0x0, Call Entry
  (Previous Disconnect Cause=0,
  Disconnect Cause=0)

*Jan 18 03:36:14.155:
  //131/2AEE8C2A801C/CCAPI/ccCallDisconnect:

  Cause Value=16, Call Entry(Responded=TRUE,
  Cause Value=16)
```

网关断开用ASR服务器建立的SIP会话

```
*Jan 18 03:36:14.159:
  //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:

Sent:

BYE sip:172.18.110.76:5060 SIP/2.0

Via: SIP/2.0/UDP 14.1.16.25:
  5060;branch=z9hG4bK971131

From: <sip:5555@14.1.16.25>;tag=E54D440-1CDB

To: <sip:asr@172.18.110.76>;tag=a99d0500
```

Date: Fri, 18 Jan 2002 03:34:57 GMT

Call-ID: 2DCAF817-AFB11D6-80D5DC30-
3585E95A@14.1.16.25

User-Agent: Cisco-SIPGateway/IOS-12.x

Max-Forwards: 70

Timestamp: 1011324974

CSeq: 102 BYE

Reason: Q.850;cause=16

Content-Length: 0

*Jan 18 03:36:14.607:

//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:

Received:

SIP/2.0 200 OK

Via: SIP/2.0/UDP 14.1.16.25:
5060;branch=z9hG4bK971131

To: <sip:asr@172.18.110.76>;tag=a99d0500

From: <sip:5555@14.1.16.25>;tag=E54D440-1CDB

Call-ID: 2DCAF817-AFB11D6-80D5DC30-
3585E95A@14.1.16.25

CSeq: 102 BYE

Contact: <sip:172.18.110.76:5060>

Content-Length: 0

[网关断开用TTS服务器建立的SIP会话](#)

*Jan 18 03:36:14.159:

//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:

Sent:

BYE sip:172.18.110.76:5060 SIP/2.0

Via: SIP/2.0/UDP 14.1.16.25:5060;branch=z9hG4bK981487

From: <sip:5555@14.1.16.25>;tag=E54D43C-1EC4

To: <sip:tts@172.18.110.76>;tag=c1160600

Date: Fri, 18 Jan 2002 03:34:57 GMT

Call-ID: 2DCA5BEF-AFB11D6-

80D3DC30-3585E95A@14.1.16.25

User-Agent: Cisco-SIPGateway/IOS-12.x

Max-Forwards: 70

Timestamp: 1011324974

CSeq: 102 BYE

Reason: Q.850;cause=16

Content-Length: 0

*Jan 18 03:36:14.215:

//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:

Received:

SIP/2.0 200 OK

Via: SIP/2.0/UDP

14.1.16.25:5060;branch=z9hG4bK981487

To: <sip:tts@172.18.110.76>;tag=c1160600

From: <sip:5555@14.1.16.25>;tag=E54D43C-1EC4

Call-ID:

2DCA5BEF-AFB11D6-80D3DC30-3585E95A@14.1.16.25

CSeq: 102 BYE

Contact: <sip:172.18.110.76:5060>

Content-Length: 0

[网关断开在ISDN旁拉的呼叫](#)

*Jan 18 03:36:14.611: ISDN Se3/0:23 Q931: TX ->

DISCONNECT pd = 8 callref = 0x805A

Cause i = 0x8090 - Normal call clearing

*Jan 18 03:36:14.623: ISDN Se3/0:23 Q931:

RX <- RELEASE pd = 8 callref = 0x005A

*Jan 18 03:36:14.623: ISDN Se3/0:23 Q931:

TX -> RELEASE_COMP pd = 8 callref = 0x805A

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