使用MRCPv2 ASR/TTS的IOS語音XML網關到CVP呼叫流

目錄

簡介

語音可擴展標籤語言(VXML)是由全球資訊網聯盟(W3C)定義的標準。 它旨在建立音訊對話，提供合成語音、口語識別、DTMF數字識別以及錄製的口語音訊。VXML伺服器和客戶端使用公認的HTTP協定交換VXML文檔/頁面。

Cisco Voice Portal(CVP)提供可通過電話訪問的智慧互動式語音響應(IVR)應用。CVP部署分為三種型別：

1. 獨立服務
2. CVP通話控制
3. 呼叫隊列和轉接

合成語音和口語/DTMF數字功能由文本到語音轉換(TTS)和自動語音識別伺服器(ASR)提供。 IOS® VXML網關通過媒體資源控制協定(MRCP)與TTS/ASR伺服器通訊。 MRCP(RFC 4463)有兩個版本，即MRCPv1(MRCP over RTSP)和MRCPv2(MRCP over SIP)。

本文檔介紹使用MRCPv2 TTS/ASR伺服器的獨立服務部署中IOS語音XML網關到CVP呼叫的呼叫流程。在CVP VXML伺服器上部署了一個示例藥房應用程式式。

必要條件

需求

本文件沒有特定需求。

採用元件

本文中的資訊係根據以下軟體和硬體版本：

- IOS VXML網關：Cisco AS5400XM，IOS 12.4(15)T1
- VXML伺服器：CVP 4.0
- ASR/TTS伺服器：倫琴多語音套件7.0

本文中的資訊是根據特定實驗室環境內的裝置所建立。文中使用到的所有裝置皆從已清除（預設）的組態來啟動。如果您的網路正在作用，請確保您已瞭解任何指令可能造成的影響。

慣例

如需文件慣例的詳細資訊，請參閱思科技術提示慣例。

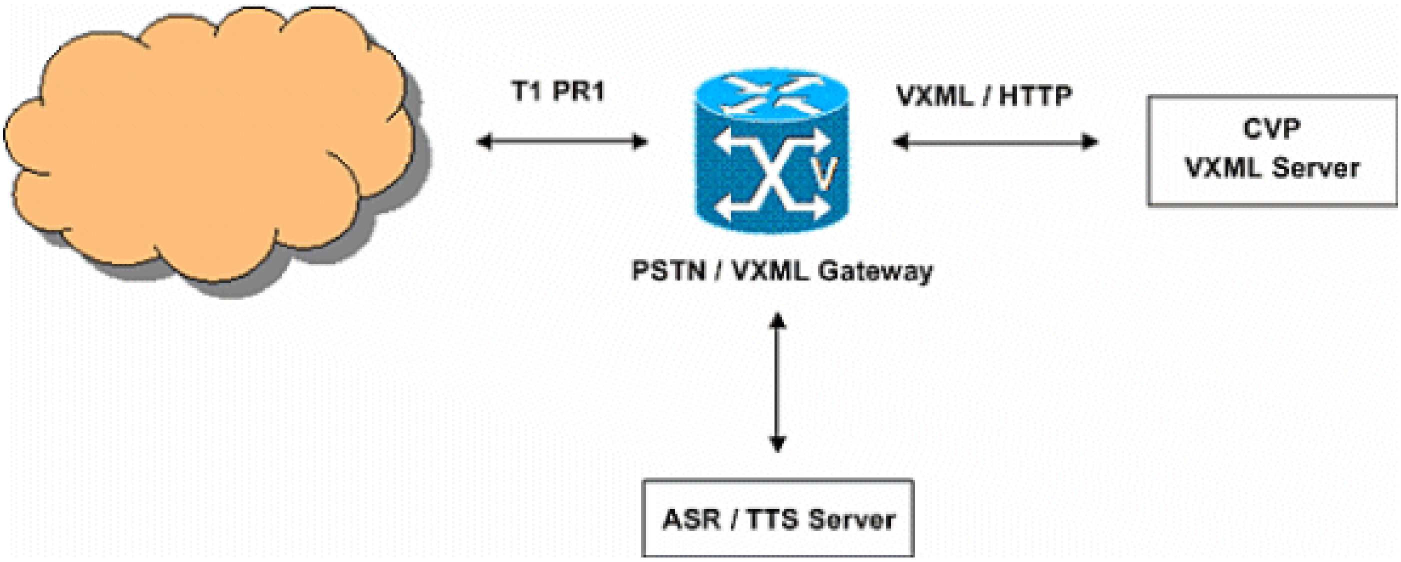
設定

本節提供用於設定本文件中所述功能的資訊。

註：使用Command Lookup Tool(僅供已註冊客戶使用)可獲取本節中使用的命令的詳細資訊。

網路圖表

本檔案會使用以下網路設定：



組態

本檔案會使用以下設定：

VXML網關配置
!--- 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 !--- CVPPrimary 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 CVPPrimaryVXMLServer
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

本節介紹此配置示例產生的呼叫流。

1. ISDN呼叫通過T1 PRI 3/0到達PSTN/VXML網關。
2. IOS網關將POTS撥號對等體1匹配為此呼叫的入站撥號對等體。
3. IOS網關將呼叫控制交給與撥號對等體1關聯的藥房服務。
4. 與藥房服務關聯的CVP VXML/TCL指令碼向VXML伺服器傳送HTTP GET請求。
5. VXML伺服器返回200 OK響應。此響應包含VXML文檔/頁面。
6. IOS網關執行VXML文檔。
7. 如果VXML文檔指定了音訊提示的URL，則IOS網關將下載音訊檔案並播放提示。
8. 如果VXML文檔指定了音訊提示的文本，則IOS網關使用撥號對等體5與tts@172.18.110.76（TTS伺服器）建立SIP會話。建立SIP會話後，使用SIP INVITE的SDP 200 OK響應中提供的TCP埠號開啟與TTS伺服器的TCP連線。此TCP連線用於在IOS網關和TTS伺服器之間交換MRCP消息，例如SPEAK、SPEAK-COMPLETE。TTS伺服器將G.711ulaw RTP音訊流傳送到網關在SIP INVITE的SDP中提供的IP地址和UDP埠號。
9. 如果VXML文檔指定網關以識別DTMF數字和/或口語，則IOS網關會使用撥號對等體6與asr@172.18.110.76（ASR伺服器）建立SIP會話。建立SIP會話後，它會使用SIP INVITE的200 OK響應的SDP中提供的TCP埠號開啟與ASR伺服器的TCP連線。此TCP連線用於在IOS網關和ASR伺服器之間交換MRCP消息，例如DEFINE GRAMMAR、COMPLETE、RECOGNITION和RECOGNITION-COMPLETE。IOS VXML網關將G.711ulaw RTP音訊流傳送到ASR在SIP 200 OK響應的SDP中提供的IP地址和UDP埠號。IOS VXML網關將PSTN使用者輸入的數字作為RTP-NTE事件傳送到ASR伺服器。
10. 執行VXML文檔後，網關將傳送HTTP POST請求（包含一組引數），如VXML文檔/頁面的<submit>標籤中所指定。
11. 對於伺服器傳送的每個VXML文檔，都會執行步驟6 - 10。
12. 當VXML應用程式完成向呼叫方提供的服務時，它會傳送一個VXML文檔，該文檔在<form>元素中僅包含<exit/>標籤。
13. IOS網關斷開與TTS和ASR伺服器建立的MRCPv2會話的連線。
14. IOS網關會斷開ISDN端的呼叫。

驗證

使用本節內容，確認您的組態是否正常運作。

輸出直譯器工具(僅供已註冊客戶使用)(OIT)支援某些**show**命令。使用OIT檢視**show**命令輸出的分析。

- **顯示呼叫活動語音簡介**

```
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
```

- **顯示呼叫活動媒體簡介**

```
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/a11F8 :
   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
```

- **Show mrcp client session active detail**

```
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
```

疑難排解

本節提供的資訊可用於對組態進行疑難排解。

Debug指令

配置IOS網關以在其日誌緩衝區中記錄調試並禁用「日誌控制檯」。

附註：使用 debug 指令之前，請先參閱有關 Debug 指令的重要資訊。

注意：以下命令用於配置網關，以便將調試儲存在網關的日誌緩衝區中：

- **service timestamps debug datetime msec**
- **服務序列**
- **無日誌控制檯**
- **logging buffered 5000000 debug**
- **清除日誌**

以下是用於對配置進行故障排除的debug命令：

- **debug isdn q931**
- **debug voip ccapi inout**
- **debug voip application vxml default**
- **debug voip application vxml dump**
- **debug ccsip message**
- **debug mrcp detail**
- **debug http client all**
- **debug voip rtp session nte named-event**

調試輸出

本節提供此呼叫流程示例的調試輸出：

1. 網關收到來自PSTN的入站呼叫。
2. 網關與入站撥號對等體1匹配。
3. 該呼叫將被轉接至藥房服務。
4. 呼叫在ISDN端連線。
5. 網關開始執行CVPSelfServiceBootstrap.vxml VoiceXML指令碼。
6. 網關向VXML伺服器傳送HTTP GET請求。
7. 網關收到來自VXML伺服器的200 OK消息。此響應的消息正文包含VXML文檔(1)。 此VXML文檔告知位於媒體伺服器中的網關播放媒體檔案Welcome-1.wav。
8. 網關向媒體伺服器傳送HTTP GET請求以下載Welcome-1.wav檔案。
9. 網關從媒體伺服器收到200 OK並接收HTTP消息正文中Welcome-1.wav的內容。
10. 網關將POST HTTP請求傳送到VXML文檔(1)的「提交」選項中定義的伺服器。
11. 網關收到其POST HTTP請求的200 OK。郵件正文包含VXML文檔(2)。 此VXML文檔告知網關播放「感謝您致電海航藥店」。 請注意，此提示需要由Text to Speech Server合成。
12. 網關按照VXML文檔(2)的「提交」選項中的定義傳送HTTP POST請求。
13. 網關收到HTTP POST請求的200 OK響應。郵件正文包含VXML文檔(3)。 此VXML文檔定義了一個選單提示，告訴呼叫者輸入1或說重新填寫2或說藥劑師。提示由文本到語音伺服器合成。使用自動語音識別器識別輸入（語音/DTMF）。
14. 網關建立用於DTMF/語音識別的語法。一旦網關與ASR伺服器建立會話，這些語法就會傳送

到ASR伺服器。

15. 網關執行撥號對等查詢以設定與文本到語音伺服器的SIP會話。出站撥號對等體6匹配。
16. 網關向TTS伺服器傳送SIP邀請。INVITE消息的SDP包含音訊流和MRCPv2應用（語音合成通道）的媒體資訊。
17. 網關執行撥號對等查詢以設定與自動語音識別伺服器的SIP會話。出站撥號對等體5已匹配。
18. 網關向ASR伺服器傳送SIP INVITE。SDP包含音訊流、DTMF中繼和MRCPv2應用（語音錄製通道）的媒體資訊。
19. 網關收到來自ASR伺服器的200 OK響應（針對SIP INVITE）。SIP INVITE消息的SDP指定了以下內容：音訊流的G711ulaw編解碼器、IP地址和RTP埠號此RTP流的direction屬性："recvonly"基於RTP-NTE的DTMF中繼網關用於與ASR伺服器建立MRCPv2會話的TCP埠號(51001)
20. 網關將SIP ACK傳送到ASR伺服器，並在網關和ASR伺服器之間建立用於自動語音識別的SIP會話。
21. 網關向ASR伺服器傳送「DEFINE-GRAMMER」MRCP請求。（此處只顯示一個請求。）
22. 網關收到其DEFINE-GRAMMAR請求的200 COMPLETE響應。
23. 網關收到來自TTS伺服器的200 OK響應（針對SIP INVITE）。SIP INVITE消息的SDP指定了以下內容：音訊流的G711ulaw編解碼器、IP地址和RTP埠號此RTP流的方向屬性："sendonly"基於RTP-NTE的DTMF中繼網關用於與TTS伺服器建立MRCPv2會話的TCP埠號(51000)
24. 網關將SIP ACK傳送到TTS伺服器，並且在網關和TTS伺服器之間建立用於文本到語音的SIP會話。
25. 網關向ASR伺服器傳送「識別」MRCP請求，以開始識別DTMF/口語。
26. ASR伺服器向網關傳送「正在進行」響應（用於識別請求）。
27. Gateway完成Welcome-1.wav媒體檔案的下載，將其儲存在快取中，並向呼叫者播放提示。
28. 網關向TTS伺服器傳送「SPEAK」 MRCP請求以播放「感謝呼叫」提示。
29. TTS伺服器向SPEAK請求傳送「進行中」響應。
30. TTS伺服器在發出「感謝呼叫」提示後傳送一則「SPEAK-COMPLETE」消息。
31. 網關向TTS伺服器傳送「SPEAK」 MRCP請求以播放「選單」提示（輸入1或說Refil/輸入2或說pharmacist）。（未顯示調試輸出。）
32. TTS伺服器傳送IN-PROGRESS， SPEAK-COMPLETE消息並完成提示播放。（未顯示調試輸出。）
33. PSTN呼叫者輸入「1」以選擇重新填寫。網關將此數字作為RTP-NTE事件傳送到ASR伺服器。
34. ASR伺服器向網關傳送「識別完成」消息，通知網關它識別到了所請求的一個事件（在本例中為數字1）。
35. 在從ASR伺服器接收到成功的識別通知後，VXML網關傳送一個HTTP POST請求，如VXML文檔的SUBMIT標籤中所指定(3)。 此POST請求通知VXML伺服器PSTN呼叫方輸入了數字1。
36. 然後VXML伺服器傳送另一個VXML文檔，要求呼叫者在此處輸入處方。（未顯示調試輸出。）
37. 網關向TTS傳送MRCP消息以發出提示。（未顯示調試輸出，但它們類似於步驟28-30。）
38. 網關向ASR傳送MRCP消息，以檢測使用者所說的4位處方編號。（未顯示調試輸出，但是它們類似於步驟25-26。）
39. ASR識別4位處方編號，並向IOS VXML網關傳送「RECOGNITION-COMPLETE」 MRCP消息。
40. 網關通過傳送HTTP POST請求將處方編號通知給VXML伺服器。（未顯示調試輸出，但是它們類似於步驟35。）
41. VXML伺服器傳送VXML頁面以收集取貨時間並通知呼叫方處方已準備好取貨。網關通過與

TTS和ASR伺服器的互動執行這些頁面。（未顯示調試輸出。）

42. VXML伺服器傳送的最終VXML文檔僅包含<form>中的<exit\>標籤。這將通知網關終止VXML會話。
43. 網關終止VXML應用程式。
44. 網關斷開與ASR伺服器建立的SIP會話的連線。
45. 網關斷開與TTS伺服器建立的SIP會話的連線。
46. 網關斷開ISDN端呼叫。

來自PSTN的入站呼叫

```
*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
```

入站撥號對等體1已匹配

```
*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
```

呼叫已轉接至藥房服務

```
*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端連線

```
*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:
```

網關開始執行CVPSelfServiceBootstrap.vxml VoiceXML指令碼

```
*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
```

網關向VXML伺服器傳送HTTP GET請求

```
*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
```

網關收到來自VXML伺服器的200 OK消息

此響應的消息正文包含VXML文檔(1)。 VXML文檔告知位於媒體伺服器中的網關播放媒體檔案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&amp;
   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>
```

網關向媒體伺服器傳送HTTP GET請求以下載Welcome-1.wav檔案

```
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
```

網關從媒體伺服器收到200 OK並接收HTTP消息正文中Welcome-1.wav的內容

```
*Jan 18 03:34:55.647:
   //127//HTTPC:/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
   F00401F0010800066163744000176700
   64617461176700FFFFFF807
   FFFFFFF80FFFFFF80F
(other hex information not shown).
```

[網關按照VXML文檔的「提交」選項中的定義向伺服器傳送POST HTTP請求(1)]

```
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
```

[網關收到其POST HTTP請求的200 OK]

郵件正文包含VXML文檔(2)。 VXML文檔告知網關播放「感謝您致電海航藥店」。 請注意，此提示需要由Text to Speech Server合成。

```
*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&amp;
   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>
```

網關按照VXML文檔提交選項中的定義傳送HTTP POST請求(2)

```
*Jan 18 03:34:55.667:
   //127//HTTPC:/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=
   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-data,
   application/octet-stream
User-Agent: Cisco-IOS-C5400/12.4
```

網關收到HTTP POST請求的200 OK響應

郵件正文包含VXML文檔(3)。 此VXML文檔定義了一個選單提示，告訴呼叫者輸入1或說重新填寫，或輸入2或說藥劑師。提示由文本到語音伺服器合成。輸入（語音/DTMF）通過自動語音識別器識別。

```
*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&amp;
   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$$$' + choice_fld + '^^^'
   + application.getElapsedTim
   (audium_element_start_time_millisecs)" />
        <assign name="audium_vxmlLog"
   expr="audium_vxmlLog
   + '|||confidence$$$' + 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>
```

[網關建立用於DTMF/語音識別的語法]

一旦網關與ASR伺服器建立會話，這些語法就會傳送到ASR伺服器。

```
*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:/mrcp_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:/mrcp_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:/mrcp_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
```

網關執行撥號對等查詢以設定與文本到語音伺服器的SIP會話

出站撥號對等體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=)
```

網關向TTS伺服器傳送SIP邀請

INVITE消息的SDP包含音訊流和MRCPv2應用（語音合成通道）的媒體資訊。

```
*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
```

網關執行撥號對等查詢以與ASR伺服器建立SIP會話

出站撥號對等體5已匹配。

```
*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/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: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=)
```

網關向ASR伺服器傳送SIP邀請

SDP包含音訊流DTMF中繼的媒體資訊。和MRCPv2應用程式（speechrecog通道）。

```
*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
```

網關從ASR伺服器收到200 OK響應（針對SIP INVITE）

1. 音訊流的G711ulaw編解碼器、IP地址和RTP埠號。
2. 此RTP流的方向屬性為「recvonly」。
3. 基於RTP-NTE的DTMF中繼。
4. 網關用於與ASR伺服器建立MRCPv2會話的TCP埠號(51001)。

```
*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
```

網關將SIP ACK傳送到ASR伺服器

ASR的SIP會話在網關和ASR伺服器之間建立。

```
*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
```

網關收到其DEFINE-GRAMMAR請求的200個完整響應

```
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
```

網關將「DEFINE-GRAMMER」MRCP請求傳送到ASR伺服器

此處只顯示一個請求。

```
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"
   mlns="http://www.w3.org/2001/06/grammar"
   xml:lang="en-us" root="root"

><rule id="root" scope="public">
   prescription</rule></grammar>
```

網關收到其DEFINE-GRAMMAR請求的200個完整響應

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

   Table=mrcpv2_socket_connect_table, Key=0:

MRCP/2.0 80 1 200 COMPLETE
```

```
Channel-Identifier: 000023B846361276@speechrecog
```

網關收到來自TTS伺服器的200 OK響應（針對SIP INVITE）

SIP INVITE消息的SDP指定了以下內容：

1. 音訊流的G711ulaw編解碼器、IP地址和RTP埠號。
2. 此RTP流的direction屬性為「sendonly」。
3. 基於RTP-NTE的DTMF中繼
4. 網關用於與TTS伺服器建立MRCPv2會話的TCP埠號(51000)。

```
*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
```

網關向TTS伺服器傳送SIP ACK

文本到語音的SIP會話在網關和TTS伺服器之間建立。

```
*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
```

網關向ASR伺服器傳送「識別」MRCP請求

```
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伺服器向網關傳送「正在進行」響應（用於識別請求）

```
MRCP/2.0 84 15 200 IN-PROGRESS

Channel-Identifier:
   000023B846361276@speechrecog
```

網關完成Welcome-1.wav媒體檔案的下載

將其儲存在快取中，並向呼叫方播放提示。

```
*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:
```

網關將「SPEAK」MRCP請求傳送到TTS伺服器以播放感謝提示

```
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伺服器傳送SPEAK請求的「進行中」響應

```
MRCP/2.0 83 1 200 IN-PROGRESS

Channel-Identifier:
   000023EC46361276@speechsynth
```

TTS伺服器在說出感謝提示後傳送「SPEAK-COMPLETE」消息

```
MRCP/2.0 141 SPEAK-COMPLETE 1 COMPLETE

Channel-Identifier:
   000023EC46361276@speechsynth

Completion-Cause: 000 normal

Speech-Marker: ""
```

PSTN呼叫者輸入「1」以選擇重新填寫

網關將此數字作為RTP-NTE事件傳送到ASR伺服器。

```
*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伺服器向網關傳送「識別完成」消息

這將通知網關它已識別其中一個請求的事件（在本例中為數字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網關收到來自ASR伺服器的成功識別通知

收到此通知後，VXML網關將傳送HTTP POST請求，如VXML文檔的SUBMIT標籤中所指定(3)。 此POST請求通知VXML伺服器PSTN呼叫方輸入了數字1。

```
*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-urlencod

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識別4位處方編號

ASR向IOS VXML網關傳送識別完成MRCP消息。

```
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伺服器傳送的最後一個VXML文檔僅包含表單中的退出標籤

這將通知網關終止VXML會話

```
*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>
```

網關終止VXML應用程式

```
*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(Responsed=TRUE,
   Cause Value=16)
```

網關斷開與ASR伺服器建立的SIP會話的連線

```
*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
```

網關斷開與TTS伺服器建立的SIP會話的連線

```
*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
```

網關斷開ISDN端呼叫

```
*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
```

相關資訊

- 語音技術支援
- 語音和整合通訊產品支援
- Cisco IP電話故障排除
- 技術支援與文件 - Cisco Systems