IOS Voice XML Gateway para fluxo de chamada CVP usando MRCPv2 ASR / TTS

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Introduction

Voice Extensible Markup Language (VXML) é um padrão definido pelo World Wide Web Consortium (W3C). Ele foi projetado para criar diálogos de áudio que fornecem fala sintetizada, reconhecimento de palavras faladas, reconhecimento de dígitos DTMF e áudio falado gravado. O servidor VXML e os clientes usam o protocolo HTTP bem conhecido para trocar documentos / páginas VXML.

O Cisco Voice Portal (CVP) oferece aplicativos inteligentes e interativos de resposta de voz (IVR) que podem ser acessados pelo telefone. Há três tipos de implementações CVP:

1. Serviço independente
2. Controle de chamada CVP
3. Fila e transferência de chamadas

A fala sintetizada e o reconhecimento de palavras faladas/funcionalidades de dígitos DTMF são fornecidos por TTS (Text-to-Speech) e ASR (Automatic Speech Recognition Servers, servidores de reconhecimento automático de voz). O IOS® VXML Gateway se comunica com o servidor TTS/ASR através do Media Resource Control Protocol (MRCP). Há duas versões de MRCP (RFC 4463), nomeadamente MRCPv1 (MRCP sobre RTSP) e MRCPv2 (MRCP sobre SIP).

Este documento descreve o fluxo de chamadas de um IOS Voice XML Gateway para a chamada CVP em uma implantação de serviço independente que usa servidores TTS/ASR MRCPv2. Um exemplo de aplicativo de farmácia foi implantado no servidor VXML do CVP.

Prerequisites

Requirements

Não existem requisitos específicos para este documento.

Componentes Utilizados

As informações neste documento são baseadas nestas versões de software e hardware:

- Gateway VXML do IOS: Cisco AS5400XM, IOS 12.4(15)T1
- Servidor VXML: CVP 4.0
- Servidor ASR / TTS: Loquendo Speech Suite 7.0

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Conventions

Consulte as Convenções de Dicas Técnicas da Cisco para obter mais informações sobre convenções de documentos.

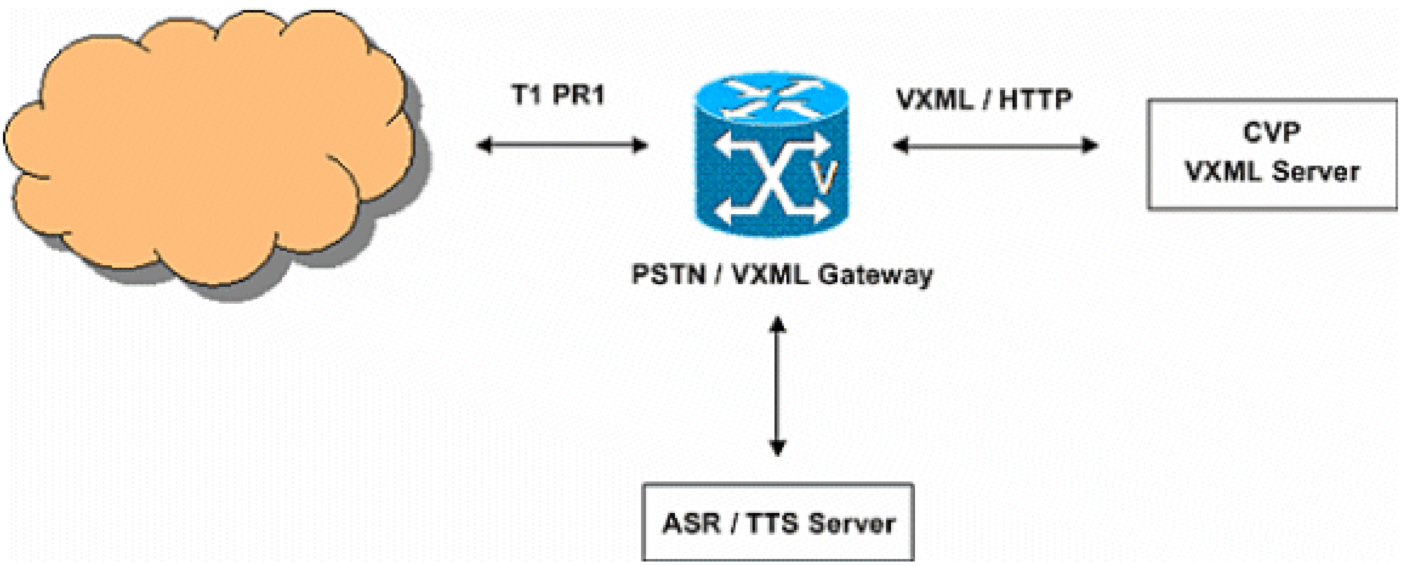
Configurar

Nesta seção, você encontrará informações para configurar os recursos descritos neste documento.

Nota: Use a Command Lookup Tool (somente clientes registrados) para obter mais informações sobre os comandos usados nesta seção.

Diagrama de Rede

Este documento utiliza a seguinte configuração de rede:



Configurações

Este documento utiliza as seguintes configurações:

Configuração do gateway VXML
!--- 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
```

Exemplo de fluxo de chamada

Esta seção descreve o fluxo de chamada que resulta deste exemplo de configuração.

1. Uma chamada ISDN chega ao Gateway PSTN/VXML através do T1 PRI 3/0.
2. O IOS Gateway corresponde ao peer de discagem POTS 1 como o peer de discagem de entrada para esta chamada.
3. O IOS Gateway transfere o controle de chamada para o serviço de farmácia associado ao dial-peer 1.
4. O script VXML / TCL do CVP associado ao serviço Pharmacy envia uma solicitação HTTP GET ao servidor VXML.
5. O servidor VXML retorna resposta 200 OK. Esta resposta contém um documento/página VXML.
6. O IOS Gateway executa o documento VXML.
7. Se o documento VXML especificar um URL para um prompt de áudio, o IOS Gateway baixará o arquivo de áudio e reproduzirá o prompt.
8. Se o documento VXML especificar um texto para um prompt de áudio, o IOS Gateway estabelece uma sessão SIP com tts@172.18.110.76 (Servidor TTS) usando o peer de discagem 5. Depois que a sessão SIP é estabelecida, ela abre uma conexão TCP com o Servidor TTS usando o número de porta TCP fornecido no SDP de 200 OK da resposta do CONVITE SIP. Essa conexão TCP é usada para trocar mensagens de MRCP como SPEAK, SPEAK-COMPLETE entre o IOS Gateway e o TTS Server.O Servidor TTS envia o fluxo de áudio RTP G.711ulaw para o endereço IP e o número da porta UDP fornecidos pelo Gateway no SDP do CONVITE SIP.
9. Se o documento VXML especificar o gateway para reconhecer dígitos DTMF e/ou palavras faladas, o IOS Gateway estabelece uma sessão SIP com asr@172.18.110.76 (servidor ASR) com dial-peer 6. Depois que a sessão SIP é estabelecida, ela abre uma conexão TCP com o Servidor ASR usando o número da porta TCP fornecido no SDP de 200 OK da resposta do CONVITE SIP. Essa conexão TCP é usada para trocar mensagens de MRCP como DEFINE GRAMMAR, COMPLETE, RECOGNIZE e RECOGNITION-COMPLETE entre o IOS Gateway e o ASR Server.O IOS VXML Gateway envia o fluxo de áudio de RTP G.711ulaw para o endereço IP e o número da porta UDP fornecidos pelo ASR no SDP da resposta SIP 200 OK. O IOS VXML Gateway envia os dígitos inseridos pelo usuário PSTN como eventos RTP-NTE para o servidor ASR.
10. Após a execução do documento VXML, o gateway envia uma solicitação HTTP POST (com um conjunto de parâmetros) conforme especificado na marca <Submit> do documento/página VXML.
11. As etapas 6 a 10 ocorrem para cada documento VXML enviado pelo servidor.
12. Quando o aplicativo VXML finaliza o serviço fornecido ao chamador, ele envia um documento VXML com apenas uma marca <exit/> no elemento <form>.
13. O IOS Gateway desconecta as sessões MRCPv2 estabelecidas com os servidores TTS e ASR.

14. O IOS Gateway desconecta a chamada no lado ISDN.

Verificar

Use esta seção para confirmar se a sua configuração funciona corretamente.

A Output Interpreter Tool (somente clientes registrados) (OIT) oferece suporte a determinados comandos show. Use a OIT para exibir uma análise da saída do comando show.

- **Mostrar resumo de voz ativa da chamada**

```
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
```

- **Mostrar resumo da mídia ativa da chamada**

```
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 ative 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
```

- **Mostrar conexões de voip rtp**

```
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
```

- **Mostrar cache de cliente http**

```
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
```

Troubleshoot

Esta seção fornece informações que podem ser usadas para o troubleshooting da sua configuração.

Comandos debug

Configure o IOS Gateway para registrar as depurações em seu buffer de registro e desabilitar o "console de registro".

Nota:Consulte Informações Importantes sobre Comandos de Depuração antes de usar comandos debug.

Observação: estes são os comandos usados para configurar o Gateway para armazenar as depurações no buffer de registro do Gateway:

- **service timestamps debug datetime msec**
- **sequência de serviço**
- **no logging console**
- **logging buffered 500000 debug**
- **clear log**

A seguir estão os comandos debug usados para solucionar problemas da configuração:

- **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 names**

Saídas de depuração

Esta seção fornece saídas de depuração para este fluxo de chamada de exemplo:

1. O gateway recebe uma chamada de entrada do PSTN.
2. O gateway corresponde ao correspondente de discagem de entrada 1.
3. A chamada é entregue ao Serviço de Farmácia.
4. A chamada é conectada no lado ISDN.
5. O gateway inicia a execução do script CVPSelfServiceBootstrap.vxml VoiceXML.
6. O gateway envia uma solicitação HTTP GET ao VXML Server.
7. O Gateway recebe uma mensagem 200 OK do VXML Server. O corpo da mensagem desta resposta contém o documento VXML (1). Este documento VXML informa ao Gateway o arquivo de mídia de reprodução chamado Welcome-1.wav localizado em um Media Server.
8. O Gateway envia uma Solicitação HTTP GET ao Servidor de Mídia para baixar o arquivo Welcome-1.wav.
9. O Gateway recebe um 200 OK do Servidor de Mídia e recebe o conteúdo de Welcome-1.wav no corpo da mensagem HTTP.
10. O Gateway envia uma Solicitação HTTP POST ao Servidor conforme definido na opção "Enviar" do Documento VXML (1).
11. O gateway recebe 200 OK para sua solicitação POST HTTP. O corpo da mensagem contém o documento VXML (2). Este documento VXML instrui o Gateway a reproduzir "Obrigado por ligar para a farmácia de áudio". Observe que esse prompt precisa ser sintetizado por um Servidor de texto para voz.
12. O Gateway envia uma solicitação HTTP POST conforme definido na opção Submit (Enviar) do documento VXML (2).
13. O Gateway recebe uma resposta 200 OK para a solicitação HTTP POST. O corpo da mensagem contém o documento VXML (3). Este documento VXML define os prompts de menu que instruem o chamador a inserir 1 ou digitar Refill, 2 ou digitar farmacêutico. Os prompts são sintetizados por um Servidor de Texto para Fala. As entradas (fala/DTMF) são reconhecidas usando um Reconhecedor de Voz Automático.
14. O gateway cria as gramáticas a serem usadas para o reconhecimento de DTMF/Fala. Essas gramáticas são enviadas ao servidor ASR assim que o Gateway estabelece uma sessão com o servidor ASR.
15. O Gateway executa uma pesquisa de peer de discagem para configurar uma sessão SIP com o Servidor de Texto para Voz. O correspondente de discagem de saída 6 é correspondido.
16. O gateway envia um CONVITE SIP para o Servidor TTS. O SDP da mensagem CONVITE contém informações de mídia para o fluxo de áudio e o aplicativo MRCPv2 (canal de sintetização de discurso).
17. O Gateway executa uma pesquisa de peer de discagem para configurar uma sessão SIP com o servidor de Reconhecimento Automático de Voz. O correspondente de discagem de saída 5 é correspondido.
18. Os gateways enviam um CONVITE SIP para o servidor ASR. O SDP contém as informações de mídia para o fluxo de áudio, retransmissão DTMF e aplicação MRCPv2 (canal de reconhecimento de discurso).

19. O Gateway recebe uma resposta 200 OK (para o CONVITE SIP) do servidor ASR. O SDP da mensagem CONVITE SIP especifica estes:O codec G711ulaw, o endereço IP e os números de porta RTP para o fluxo de áudioO atributo de direção deste fluxo de RTP: "recvonly"O relé DTMF baseado em RTP-NTEO número da porta TCP (51001) a ser usado pelo Gateway para estabelecer uma sessão MRCPv2 com o servidor ASR
20. O Gateway envia ACK SIP para o servidor ASR, e a sessão SIP para o Reconhecimento Automático de Voz é estabelecida entre o Gateway e o servidor ASR.
21. O gateway envia uma solicitação de MRCP "DEFINE-GRAMMER" ao servidor ASR. (Apenas uma solicitação é mostrada aqui.)
22. O Gateway recebe uma resposta 200 COMPLETE para sua solicitação DEFINE-GRAMMAR.
23. O Gateway recebe uma resposta 200 OK (para o CONVITE SIP) do servidor TTS. O SDP da mensagem CONVITE SIP especifica estes:O codec G711ulaw, o endereço IP e os números de porta RTP para o fluxo de áudioO atributo de direção deste fluxo RTP:"sendonly"O relé DTMF baseado em RTP-NTEO número da porta TCP (51000) a ser usado pelo Gateway para estabelecer uma sessão MRCPv2 com o servidor TTS
24. O Gateway envia ACK SIP para o Servidor TTS e a sessão SIP para o Texto para Voz é estabelecida entre o Gateway e o servidor TTS.
25. O Gateway envia uma solicitação de MRCP "RECONHECIMENTO" ao servidor ASR para iniciar o reconhecimento de DTMF / palavras faladas.
26. O servidor ASR envia uma resposta "EM ANDAMENTO" (para solicitação de RECONHECIMENTO) ao Gateway.
27. O gateway conclui o download do arquivo de mídia Welcome-1.wav, o armazena no cache e reproduz o prompt para o chamador.
28. O Gateway envia uma solicitação de MRCP "SPEAK" ao TTS Server para reproduzir o prompt "Obrigado por ligar".
29. O Servidor TTS envia uma resposta "EM ANDAMENTO" para a solicitação SPEAK.
30. O TTS Server envia uma mensagem "SPEAK-COMPLETE" depois de falar no prompt "Obrigado por ligar".
31. O Gateway envia uma solicitação de MRCP "SPEAK" ao TTS Server para reproduzir o prompt "Menu" (Insira 1 ou Diga Refil/Enter 2 ou Diga farmacêutico). (As saídas de depuração não são exibidas.)
32. O servidor TTS envia uma mensagem IN-PROGRESS, SPEAK-COMPLETE e termina de reproduzir o prompt. (As saídas de depuração não são exibidas.)
33. O chamador PSTN digita "1" para escolher Refill. O gateway envia esse dígito como um evento RTP-NTE para o servidor ASR.
34. O Servidor ASR envia uma mensagem "RECONHECIMENTO COMPLETO" ao Gateway para notificar ao gateway que ele reconheceu um dos eventos solicitados (nesse caso, dígito 1).
35. Depois de receber uma notificação de reconhecimento bem-sucedida do servidor ASR, o Gateway VXML envia uma solicitação HTTP POST conforme especificado na tag SUBMIT do documento VXML (3). Essa solicitação POST informa ao servidor VXML que o dígito 1 foi inserido pelo chamador PSTN.
36. O servidor VXML envia outro documento VXML que solicita que o chamador insira a receita aqui. (As saídas de depuração não são exibidas.)
37. O Gateway envia a mensagem MRCP ao TTS para falar os avisos. (As saídas de depuração não são mostradas, mas são semelhantes às etapas 28-30.)
38. O Gateway envia a mensagem MRCP ao ASR para detectar o número de receita de 4

dígitos falado pelo usuário. (As saídas de depuração não são mostradas, mas são semelhantes às etapas 25-26.)

39. O ASR reconhece o número de prescrição de 4 dígitos e envia uma mensagem de MRCP "RECONHECITION-COMPLETE" para o IOS VXML Gateway.
40. O Gateway informa o número da receita ao servidor VXML enviando a solicitação HTTP POST. (As saídas de depuração não são mostradas, mas são semelhantes à etapa 35.)
41. O servidor VXML envia páginas VXML para coletar o tempo de coleta e informar ao chamador que a receita estará pronta para coleta. O Gateway executa essas páginas por interações com o servidor TTS e ASR. (As saídas de depuração não são exibidas.)
42. O documento VXML final enviado pelo servidor VXML contém apenas a marca <exit\> no <form>. Isso instrui o Gateway a encerrar a sessão VXML.
43. O gateway encerra o aplicativo VXML.
44. O gateway desconecta a sessão SIP estabelecida com o servidor ASR.
45. O gateway desconecta a sessão SIP estabelecida com o Servidor TTS.
46. O gateway desconecta a chamada no lado ISDN.

Chamada de entrada do PSTN

```
*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
```

O correspondente de discagem de entrada 1 é correspondente

```
*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
```

A chamada é entregue para o serviço de farmácia

```
*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
```

A chamada é conectada no lado ISDN

```
*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:
```

O gateway inicia a execução do script VoiceXML CVPSelfServiceBootstrap.vxml

```
*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
```

O Gateway envia uma solicitação HTTP GET ao VXML Server

```
*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
```

Gateway Recebe uma mensagem 200 OK do VXML Server

O corpo da mensagem desta resposta contém um documento VXML (1). O documento VXML informa ao Gateway o arquivo de mídia de reprodução chamado Welcome-1.wav localizado em um Media Server.

```
*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>
```

O Gateway envia uma solicitação HTTP GET ao Media Server para fazer download do arquivo Welcome-1.wav

```
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
```

O Gateway recebe um 200 OK do Servidor de mídia e recebe o conteúdo do arquivo Welcome-1.wav no corpo da mensagem HTTP

```
*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
   F00401F001080006661637440001767 00
   64617461176700FFFFFF807
   FFFFFFF80FFFFFF80F
(other hex information not shown).
```

O Gateway envia uma solicitação HTTP POST ao servidor conforme definido na opção "Enviar" do documento VXML (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
```

O Gateway recebe um 200 OK para sua solicitação de HTTP POST

O corpo da mensagem contém o documento VXML (2). O documento VXML instrui o Gateway a reproduzir "Obrigado por ligar para a farmácia de áudio". Observe que esse prompt precisa ser sintetizado por um Servidor de texto para voz.

```
*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>
```

O Gateway envia uma solicitação de POST HTTP conforme definido na opção de envio do documento VXML (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
```

[O Gateway recebe uma resposta 200 OK para a solicitação HTTP POST]

O corpo da mensagem contém o documento VXML (3). Este documento VXML define os prompts de menu que instruem o chamador a inserir 1 ou dizer Refill, ou a digitar 2 ou dizer farmacêutico. Os prompts são sintetizados por um Servidor de Texto para Fala. As entradas (fala/DTMF) são reconhecidas com um Reconhecedor de Voz Automático.

```
*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>
```

O gateway cria os grampos a serem usados para reconhecimento de voz/DTMF

Essas gramáticas são enviadas ao servidor ASR assim que o Gateway estabelece uma sessão com o servidor ASR.

```
*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
```

O gateway realiza uma pesquisa de peer de discagem para configurar uma sessão SIP com o servidor de texto a voz

O correspondente de discagem de saída 6 é correspondido.

```
*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=)
```

O gateway envia um CONVITE SIP para o servidor TTS

O SDP da mensagem CONVITE contém informações de mídia para o fluxo de áudio e o aplicativo MRCPv2 (canal de sintetização de discurso).

```
*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
```

O gateway realiza uma pesquisa de peer de discagem para configurar uma sessão SIP com o servidor ASR

O correspondente de discagem de saída 5 é correspondido.

```
*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=)
```

Os gateways enviam um CONVITE SIP para o servidor ASR

O SDP contém as informações de mídia para o fluxo de áudio, retransmissão DTMF. e MRCPv2 Application (canal de reconhecimento de fala).

```
*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
```

O gateway recebe uma resposta 200 OK (para o CONVITE SIP) do servidor ASR

1. Codec G711ulaw, endereço IP e números de porta RTP para o fluxo de áudio.
2. O atributo de direção deste fluxo RTP é "recvonly".
3. Relé DTMF baseado em RTP-NTE.
4. Número da porta TCP (51001) a ser usado pelo Gateway para estabelecer uma sessão MRCPv2 com o servidor ASR.

```
*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
```

O gateway envia ACK SIP para o servidor ASR

A sessão SIP para o ASR é estabelecida entre o Gateway e o servidor ASR.

```
*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
```

O gateway envia a solicitação MRCP "DEFINE-GRAMMER" ao servidor ASR

Apenas uma solicitação é mostrada aqui.

```
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>
```

O Gateway recebe uma resposta 200 COMPLETA para sua solicitação DEFINE-GRAMMAR

```
*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
```

O Gateway recebe uma resposta 200 OK (para o CONVITE SIP) do servidor TTS

O SDP da mensagem CONVITE SIP especifica estes:

1. Codec G711ulaw, endereço IP e números de porta RTP para o fluxo de áudio.
2. O atributo de direção deste fluxo RTP é "sendonly".
3. Relé DTMF baseado em RTP-NTE
4. Número da porta TCP (51000) a ser usado pelo Gateway para estabelecer uma sessão MRCPv2 com o servidor TTS.

```
*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
```

O gateway envia ACK SIP para o servidor TTS

A sessão SIP do Text-to-Speech é estabelecida entre o Gateway e o servidor TTS.

```
*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
```

O gateway envia uma solicitação MRCP "RECONHECIDO" ao servidor ASR

```
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
```

O servidor ASR envia resposta "EM ANDAMENTO" (para solicitação de RECONHECIMENTO) ao gateway

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

Channel-Identifier:
   000023B846361276@speechrecog
```

O Gateway conclui o download do arquivo de mídia Welcome-1.wav

Ele o armazena no cache e reproduz o prompt para o chamador.

```
*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:
```

O gateway envia a solicitação MRCP "SPEAK" ao servidor TTS para reproduzir o prompt de agradecimento

```
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>
```

O Servidor TTS envia a Resposta "EM PROGRESSO" para a Solicitação SPEAK

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

Channel-Identifier:
   000023EC46361276@speechsynth
```

O servidor TTS envia a mensagem "SPEAK-COMPLETE" depois de falar com o prompt de agradecimento

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

Channel-Identifier:
   000023EC46361276@speechsynth

Completion-Cause: 000 normal
```

```
Speech-Marker: ""
```

O chamador PSTN digita "1" para escolher Refill

O gateway envia esse dígito como um evento RTP-NTE para o servidor ASR.

```
*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>>>
```

O servidor ASR envia uma mensagem "RECONHECITION-COMPLETE" ao gateway

Isso notifica o gateway de que reconheceu um dos eventos solicitados (nesse caso, dígito 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>
```

O gateway VXML recebe uma notificação de reconhecimento bem-sucedida do servidor ASR

Após o recebimento desta notificação, o Gateway VXML envia uma solicitação HTTP POST conforme especificado na tag SUBMIT do documento VXML (3). Essa solicitação POST informa ao servidor VXML que o dígito 1 foi inserido pelo chamador PSTN.

```
*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
```

O ASR reconhece o número de assinatura de 4 dígitos

O ASR envia uma mensagem de MRCP CONCLUÍDO DE RECONHECIMENTO para o IOS VXML Gateway.

```
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
```

[O último documento VXML enviado pelo VXML Server contém apenas a etiqueta de saída no formulário]()

Isso instrui o Gateway a encerrar a sessão VXML

```
*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>
```

O gateway encerra o aplicativo VXML

```
*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)
```

O gateway desconecta a sessão SIP estabelecida com o servidor ASR

```
*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
```

O gateway desconecta a sessão SIP estabelecida com o servidor TTS

```
*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
```

O gateway desconecta a chamada no lado ISDN

```
*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
```

Informações Relacionadas

- Suporte à Tecnologia de Voz
- Suporte aos produtos de Voz e Comunicações Unificadas
- Troubleshooting da Telefonia IP Cisco
- Suporte Técnico e Documentação - Cisco Systems