Gateway do IOS Voice XML ao fluxo de chamadas CVP usando MRCPv2 ASR/TTS

Índice

Introdução

O linguagem de marcação extensível da Voz (VXML) é um padrão definido pelo consórcio da world wide web (W3C). É projetado criar os diálogos audio que fornecem o discurso sintetizado, reconhecimento das palavras dita, reconhecimento dos dígitos de DTMF, e o áudio falado gravado. O server e os clientes VXML usam o protocolo HTTP conhecido para trocar originais/páginas VXML.

Cisco exprime o portal (CVP) entrega os aplicativos inteligentes e da resposta de voz interativa (IVR) que podem ser alcançados sobre o telefone. Há três tipos de implementações CVP:

1. Serviço autônomo
2. Controle de chamadas CVP
3. Fila e transferência do atendimento

O discurso sintetizado e o reconhecimento das palavras dita/funcionalidades dos dígitos de DTMF são fornecidos pelos server textos a expressão (TTS) e do reconhecimento de discurso automático (ASR). O gateway IOS® VXML comunica-se com o server TTS/ASR com o protocolo de controle dos recursos de mídia (MRCP). Há duas versões de MRCP (RFC 4463), a saber MRCPv1 (MRCP sobre o RTSP) e MRCPv2 (MRCP sobre o SORVO).

Este original descreve o fluxo de chamadas de um gateway do IOS Voice XML ao atendimento CVP em uma distribuição de serviço autônoma que use server MRCPv2 TTS/ASR. Um aplicativo da farmácia da amostra foi distribuído no server CVP VXML.

Pré-requisitos

Requisitos

Não existem requisitos específicos para este documento.

Componentes Utilizados

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

- Gateway IO VXML: Cisco AS5400XM, IO 12.4(15)T1
- Server VXML: CVP 4.0
- Server ASR/TTS: Série 7.0 do discurso de Loquendo

As informações neste documento foram criadas a partir de dispositivos em um ambiente de laboratório específico. Todos os dispositivos utilizados neste documento foram iniciados com uma configuração (padrão) inicial. Se a sua rede estiver ativa, certifique-se de que entende o impacto potencial de qualquer comando.

Convenções

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

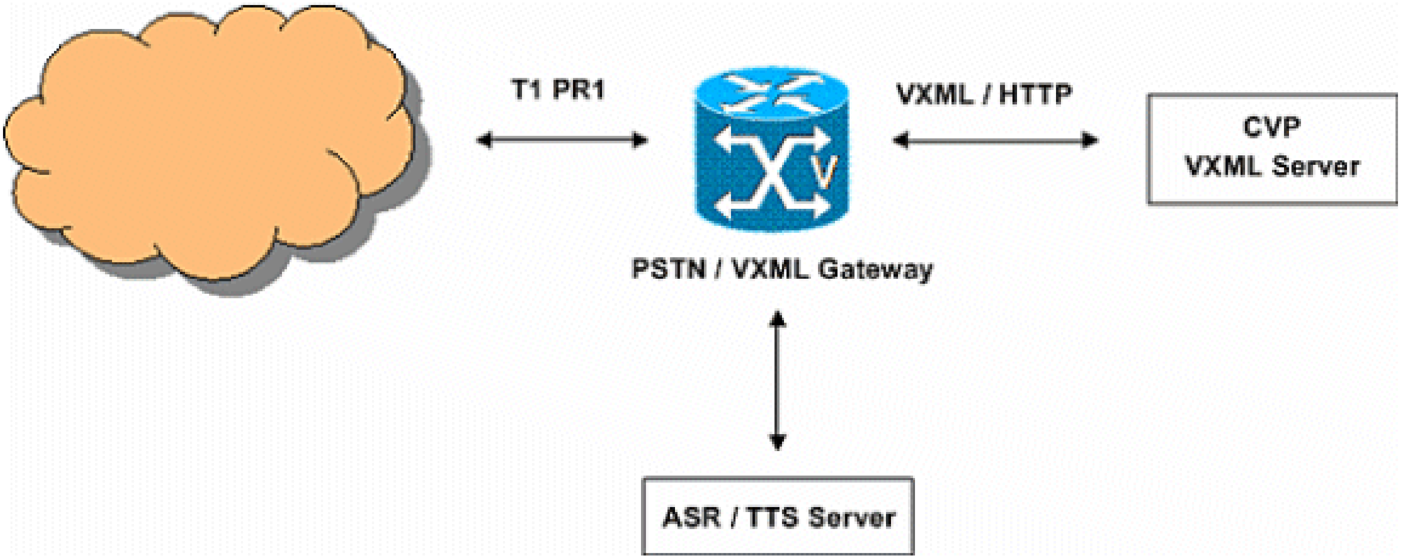
Configurar

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 de 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 chamadas

Esta seção descreve o fluxo de chamadas esse resultados deste exemplo de configuração.

1. Uma chamada ISDN chega no gateway PSTN/VXML através do T1 PRI 3/0.
2. O Gateway de IOS combina o POTS dial peer 1 como o dial peer de entrada para este atendimento.
3. O Gateway de IOS entrega fora do Controle de chamadas ao serviço da farmácia que é associado ao dial-peer 1.
4. O script CVP VXML/TCL associado com o serviço da farmácia envia um pedido HTTP GET ao server VXML.
5. O server VXML retorna uma resposta de 200 APROVAÇÕES. Esta resposta contém um original/página VXML.
6. O Gateway de IOS executa o original VXML.
7. Se o original VXML especifica uma URL para um prompt de áudio, o Gateway de IOS transfere o arquivo de áudio e joga a alerta.
8. Se o original VXML especifica um texto para um prompt de áudio, o Gateway de IOS estabelece uma sessão do SORVO com tts@172.18.110.76 (server TTS) que usa o dial-peer 5. Depois que a sessão do SORVO é estabelecida, abre uma conexão de TCP ao server TTS que usa o número de porta de TCP fornecido no SDP de uma resposta de 200 APROVAÇÕES do SORVO CONVIDA. Esta conexão de TCP é usada para trocar mensagens MRCP como FALA, SPEAK-COMPLETE entre o Gateway de IOS e o server TTS.O server TTS envia o fluxo de áudio G.711ulaw RTP ao endereço IP de Um ou Mais Servidores Cisco ICM NT e o número de porta UDP fornecido pelo gateway no SDP do SORVO CONVIDA.
9. Se o original VXML especifica o gateway para reconhecer dígitos de DTMF e/ou palavras dita, o Gateway de IOS estabelece uma sessão do SORVO com asr@172.18.110.76 (server ASR) com dial-peer 6. Depois que a sessão do SORVO é estabelecida, abre uma conexão

de TCP ao server ASR que usa o número de porta de TCP fornecido no SDP de uma resposta de 200 APROVAÇÕES do SORVO CONVIDA. Esta conexão de TCP é usada para trocar mensagens MRCP como DEFINE A GRAMÁTICA, TERMINA-A, RECONHECE-A, e RECOGNITION-COMPLETE entre o Gateway de IOS e o server ASR.O gateway IO VXML envia o fluxo de áudio G.711ulaw RTP ao endereço IP de Um ou Mais Servidores Cisco ICM NT e ao número de porta UDP fornecidos pelo ASR no SDP da resposta da APROVAÇÃO do SORVO 200. O gateway IO VXML envia os dígitos incorporados pelo usuário PSTN como eventos RTP-NTE ao server ASR.

10. Após a execução do original VXML, o gateway envia um pedido do CARGO HTTP (com um conjunto de parâmetro) como especificado na etiqueta do <submit> do original/página VXML.
11. As etapas 6 – 10 ocorrem para cada original VXML enviado pelo server.
12. Quando o aplicativo VXML termina o serviço proporcionado ao chamador, envia um original VXML com apenas uma etiqueta <exit/> dentro do elemento do <form>.
13. O Gateway de IOS desliga as sessões MRCPv2 estabelecidas com os server TTS e ASR.
14. O Gateway de IOS desliga chamar o lado ISDN.

Verificar

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

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

- **Show call ative voice brief**

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

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

- **Mostre a atendimento o resumo dos media ativos**

```
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
```

- **Mostre o detalhe do active da sessão cliente do mrcp**

```
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
```

- **Mostre conexões do rtp do voip**

```
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
```

- **Mostre o 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
```

Troubleshooting

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

Comandos debug

Configurar o Gateway de IOS para registrar debuga em seu logging buffer e desabilitam o “console de registro”.

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

Nota: Estes são os comandos usados para configurar o gateway a fim armazenar debugam no logging buffer do gateway:

- **service timestamps debug datetime msec**
- **preste serviços de manutenção à sequência**
- **nenhum console de registro**
- **registrando 5000000 protegidos debugar**
- **cancele o log**

Os seguintes são os comandos debug usados para pesquisar defeitos a configuração:

- **debug isdn q931**
- **debug voip ccapi inout**
- **debugar o padrão do vxml do aplicativo do voip**
- **debugar a descarga do vxml do aplicativo do voip**
- **debugar o mensagem de ccsip**
- **debugar o detalhe do mrcp**
- **debugar o cliente todo HTTP**
- **debugar o Nomeado-evento do nte da sessão do rtp do voip**

Saídas de depuração

Esta seção fornece resultados do debug para este fluxo de chamadas da amostra:

1. O gateway recebe uma chamada recebida do PSTN.
2. O gateway combina o dial peer de entrada 1.
3. O atendimento é entregado fora ao serviço da farmácia.
4. O atendimento obtém conectado no lado ISDN.
5. O gateway começa a execução do script do VoiceXML CVPSelfServiceBootstrap.vxml.
6. O gateway envia um pedido HTTP GET ao server VXML.
7. O gateway recebe uma mensagem de 200 APROVAÇÕES do server VXML. O corpo da mensagem desta resposta contém o original VXML (1). Este original VXML diz o arquivo de media Welcome-1.wav chamado do jogo do gateway situado em um servidor de mídia.
8. O gateway envia um pedido HTTP GET ao servidor de mídia transferir o arquivo Welcome-1.wav.
9. O gateway recebe uma APROVAÇÃO 200 do servidor de mídia e recebe os índices do Welcome-1.wav no corpo da mensagem HTTP.
10. O gateway envia um pedido do HTTP do CARGO ao server como definido no "submete" a opção do original VXML (1).
11. O gateway recebe a APROVAÇÃO 200 para seu pedido do HTTP do CARGO. O corpo da mensagem contém o original VXML (2). Este original VXML diz o gateway para jogar "agradece-lhe chamando a farmácia de Audium." Note que esta alerta precisa de ser sintetizada por um server texto a expressão.
12. O gateway envia um pedido do CARGO HTTP como definido na opção da submissão do original VXML (2).
13. O gateway recebe uma resposta de 200 APROVAÇÕES para o pedido do CARGO HTTP. O corpo da mensagem contém o original VXML (3). Este original VXML define alertas de um menu que diz o chamador para incorporar 1 ou dizer o reenchimento, 2 ou para dizer o farmacêutico. As alertas são sintetizadas por um server texto a expressão. As entradas

(discurso/DTMF) são reconhecidas usando um identificador automático do discurso.

14. O gateway cria as gramáticas a ser usadas para o DTMF/reconhecimento de discurso. Estas gramáticas estão enviadas então ao server ASR uma vez que o gateway estabelece uma sessão com o server ASR.
15. O gateway executa uma consulta do dial-peer para setup uma sessão do SORVO com o server texto a expressão. O dial peer de saída 6 é combinado.
16. O gateway envia um SORVO CONVIDA ao server TTS. O SDP do mensagem INVITE contém a informação dos media para o fluxo de áudio e o aplicativo MRCPv2 (canal do speechsynth).
17. O gateway executa uma consulta do dial-peer para setup uma sessão do SORVO com o server do reconhecimento de discurso automático. O dial peer de saída 5 é combinado.
18. Os gateways enviam um SORVO CONVIDAM ao server ASR. O SDP contém a informação dos media para o fluxo de áudio, o relé DMTF e o aplicativo MRCPv2 (canal do speechrecog).
19. O gateway recebe uma resposta de 200 APROVAÇÕES (para o SORVO CONVIDE) do server ASR. O SDP do mensagem INVITE do SORVO especifica estes:O codec G711ulaw, o endereço IP de Um ou Mais Servidores Cisco ICM NT, e os números de porta RTP para o fluxo de áudioO atributo do sentido deste córrego RTP: “recvonly”O RTP-NTE baseou o relé DMTFO número de porta de TCP (51001) a ser usado pelo gateway para estabelecer uma sessão MRCPv2 com server ASR
20. O gateway envia o SORVO ACK ao server ASR, e a sessão do SORVO para o reconhecimento de discurso automático obtém estabelecida entre o gateway e o server ASR.
21. O gateway envia um pedido MRCP “DEFINE-GRAMMER” ao server ASR. (Apenas um pedido é mostrado aqui.)
22. O gateway recebe uma resposta 200 COMPLETA para seu pedido DEFINE-GRAMMAR.
23. O gateway recebe uma resposta de 200 APROVAÇÕES (para o SORVO CONVIDE) do server TTS. O SDP do mensagem INVITE do SORVO especifica estes:O codec G711ulaw, o endereço IP de Um ou Mais Servidores Cisco ICM NT e os números de porta RTP para o fluxo de áudioO atributo do sentido deste córrego RTP: “sendonly”O RTP-NTE baseou o relé DMTFO número de porta de TCP (51000) a ser usado pelo gateway para estabelecer uma sessão MRCPv2 com server TTS
24. O gateway envia o SORVO ACK ao server TTS, e a sessão do SORVO para o texto a expressão obtém estabelecida entre o gateway e o server TTS.
25. O gateway envia “RECONHECE” o pedido MRCP ao server ASR começar o reconhecimento do DTMF/palavras dita.
26. O server ASR envia uma resposta “EM ANDAMENTO” (para o pedido RECOGNIZE) ao gateway.
27. O gateway termina a transferência do arquivo de media Welcome-1.wav, armazena-a no esconderijo, e joga-o a alerta ao chamador.
28. O gateway envia “FALA” o pedido MRCP ao server TTS jogar a “Agradecer-Você-para-chamada” da alerta.
29. O server TTS envia uma resposta “EM ANDAMENTO” ao pedido do DISCURSO.
30. O server TTS envia uma mensagem “SPEAK-COMPLETE” depois que falou a “Agradecer-você-para-chamada” da alerta.
31. O gateway envia “FALA” o pedido MRCP ao server TTS jogar a alerta do “menu” (incorpore 1 ou diga que Refil/incorpora 2 ou diga o farmacêutico). (Os resultados do debug não são mostrados.)

32. O server TTS envia uma mensagem EM ANDAMENTO, SPEAK-COMPLETE e os revestimentos que jogam a alerta. (Os resultados do debug não são mostrados.)
33. O chamador de PSTN incorpora “1" para escolher o reenchimento. O gateway envia este dígito como um evento RTP-NTE ao server ASR.
34. O server ASR envia uma mensagem “RECOGNITION-COMPLETE” ao gateway para notificar o gateway que reconheceu um dos eventos pedidos (neste caso dígito 1).
35. Depois que recebe uma notificação bem sucedida do reconhecimento do server ASR, o gateway VXML envia um pedido do CARGO HTTP como especificado na etiqueta da SUBMISSÃO do original VXML (3). Este pedido do CARGO informa o server VXML que o dígito 1 esteve incorporado pelo chamador de PSTN.
36. O server VXML envia então um outro original VXML que peça que o chamador incorpore a prescrição aqui. (Os resultados do debug não são mostrados.)
37. O gateway envia a mensagem MRCP ao TTS para falar as alertas. (Os resultados do debug não são mostrados, mas são similares às etapas 28-30.)
38. O gateway envia a mensagem MRCP ao ASR para detectar o número da prescrição de 4 dígitos falado pelo usuário. (Os resultados do debug não são mostrados, mas são similares às etapas 25-26.)
39. O ASR reconhece o número da prescrição de 4 dígitos e envia uma mensagem MRCP “RECOGNITION-COMPLETE” ao gateway IO VXML.
40. O gateway informa o número da prescrição ao server VXML enviando o pedido do CARGO HTTP. (Os resultados do debug não são mostrados, mas são similares a etapa 35.)
41. O server VXML envia páginas VXML para recolher o tempo de recolhimento e para informar o chamador que a prescrição estará pronta para o recolhimento. O gateway executa estas páginas por interações com o server TTS e ASR. (Os resultados do debug não são mostrados.)
42. O original final VXML enviado pelo server VXML contém apenas o <exit \ > etiqueta no <form>. Isto diz o gateway para terminar a sessão VXML.
43. O gateway termina o aplicativo VXML.
44. O gateway desliga a sessão do SORVO estabelecida com o server ASR.
45. O gateway desliga a sessão do SORVO estabelecida com o server TTS.
46. O gateway desliga chamar o lado ISDN.

Chamada recebida 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 dial peer de entrada 1 é combinado

```
*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
```

O atendimento é entregado fora ao serviço da 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
```

O atendimento obtém conectado 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 começa a execução do script do 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 um pedido HTTP GET ao server VXML

```
*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
```

O gateway recebe uma mensagem de 200 APROVAÇÕES do server VXML

O corpo da mensagem desta resposta contém um original VXML (1). O original VXML diz o arquivo de media Welcome-1.wav chamado do jogo do gateway situado em um servidor de mídia.

```
*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 um pedido HTTP GET ao servidor de mídia transferir o 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 uma APROVAÇÃO 200 do servidor de mídia e recebe os índices do 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
   F00401F00108000666163744000176700
   64617461176700FFFFFF807
   FFFFFFF80FFFFFF80F
(other hex information not shown).
```

O gateway envia um pedido do HTTP do CARGO ao server como definido no “submete” a opção do original 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 uma APROVAÇÃO 200 para seu pedido do HTTP do CARGO

O corpo da mensagem contém o original VXML (2). O original VXML diz o gateway para jogar “agradece-lhe chamando a farmácia de Audium.” Note que esta alerta precisa de ser sintetizada por um server texto a expressão.

```
*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 um pedido do CARGO HTTP como definido na opção da submissão do original 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 de 200 APROVAÇÕES para o pedido do CARGO HTTP

O corpo da mensagem contém o original VXML (3). Este original VXML define alertas de um menu que diz o chamador para incorporar 1 ou dizer o reenchimento, ou para incorporar 2 ou dizer o farmacêutico. As alertas são sintetizadas por um server texto a expressão. As entradas (discurso/DTMF) são reconhecidas com um identificador automático do discurso.

```
*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 as gramáticas a ser usadas para o DTMF/reconhecimento de discurso

Estas gramáticas estão enviadas então ao server ASR uma vez que o gateway estabelece uma sessão com o server 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 executa uma consulta do dial-peer para setup uma sessão do SORVO com o server texto a expressão

O dial peer de saída 6 é combinado.

```
*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 SORVO CONVIDA ao server TTS

O SDP do mensagem INVITE contém a informação dos media para o fluxo de áudio e o aplicativo MRCPv2 (canal do speechsynth).

```
*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 executa uma consulta do dial-peer para estabelecer uma sessão do SORVO com o server ASR

O dial peer de saída 5 é combinado.

```
*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 SORVO CONVIDAM ao server ASR

O SDP contém a informação dos media para o fluxo de áudio, relé DMTF. e aplicativo MRCPv2 (canal do speechrecog).

```
*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 de 200 APROVAÇÕES (para o SORVO CONVIDE) do server ASR

1. Codec G711ulaw, endereço IP de Um ou Mais Servidores Cisco ICM NT e números de porta RTP para o fluxo de áudio.
2. O atributo do sentido deste córrego RTP é “recvonly”.
3. RTP-NTE baseou o relé DMTF.
4. Número de porta de TCP (51001) a ser usado pelo gateway para estabelecer uma sessão MRCPv2 com server 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 o SORVO ACK ao server ASR

A sessão do SORVO para o ASR obtém estabelecida entre o gateway e o server 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 o pedido MRCP “DEFINE-GRAMMER” ao server ASR

Apenas um pedido é mostrado 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 seu pedido 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 de 200 APROVAÇÕES (para o SORVO CONVIDE) do server TTS

O SDP do mensagem INVITE do SORVO especifica estes:

1. Codec G711ulaw, endereço IP de Um ou Mais Servidores Cisco ICM NT e números de porta RTP para o fluxo de áudio.
2. O atributo do sentido deste córrego RTP é “sendonly”.
3. RTP-NTE baseou o relé DMTF

4. Número de porta de TCP (51000) a ser usado pelo gateway para estabelecer uma sessão MRCPv2 com server 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 o SORVO ACK ao server TTS

A sessão do SORVO para o texto a expressão obtém estabelecida entre o gateway e o server 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 “RECONHECE” o pedido MRCP ao server 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 server ASR envia a resposta “EM ANDAMENTO” (para RECONHEÇA o pedido) ao gateway]

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

Channel-Identifier:
   000023B846361276@speechrecog
```

[O gateway termina a transferência do arquivo de media Welcome-1.wav]

Armazena-o no esconderijo e joga-o a alerta ao 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 “FALA” o pedido MRCP ao server TTS jogar a alerta obrigado

```
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 server TTS envia a resposta “EM ANDAMENTO " para o pedido do DISCURSO

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

Channel-Identifier:
   000023EC46361276@speechsynth
```

O server TTS envia a mensagem “SPEAK-COMPLETE” depois que falou a alerta obrigado

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

Channel-Identifier:
   000023EC46361276@speechsynth

Completion-Cause: 000 normal

Speech-Marker: ""
```

O chamador de PSTN incorpora “1" para escolher o reenchimento

O gateway envia este dígito como um evento RTP-NTE ao server 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 server ASR envia uma mensagem “RECOGNITION-COMPLETE” ao gateway

Isto notifica o gateway que reconheceu um dos eventos pedidos (neste 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 bem sucedida do reconhecimento do server ASR

Após o recibo desta notificação, o gateway VXML envia um pedido do CARGO HTTP como especificado na etiqueta da SUBMISSÃO do original VXML (3). Este pedido do CARGO informa o server VXML que o dígito 1 esteve incorporado pelo chamador de 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 da prescrição do 4-dígito

O ASR envia uma mensagem RECOGNITION-COMPLETE MRCP ao gateway IO VXML.

```
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 original VXML enviado pelo server VXML contém apenas a etiqueta da saída no formulário

Isto diz o gateway para terminar 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 termina 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 desliga a sessão do SORVO estabelecida com o server 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 desliga a sessão do SORVO estabelecida com o server 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 desliga chamar o 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 ao Produto de Voz e Comunicações Unificadas
- Troubleshooting da Telefonia IP Cisco
- Suporte Técnico e Documentação - Cisco Systems