

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

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[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 documentos/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 documento 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.

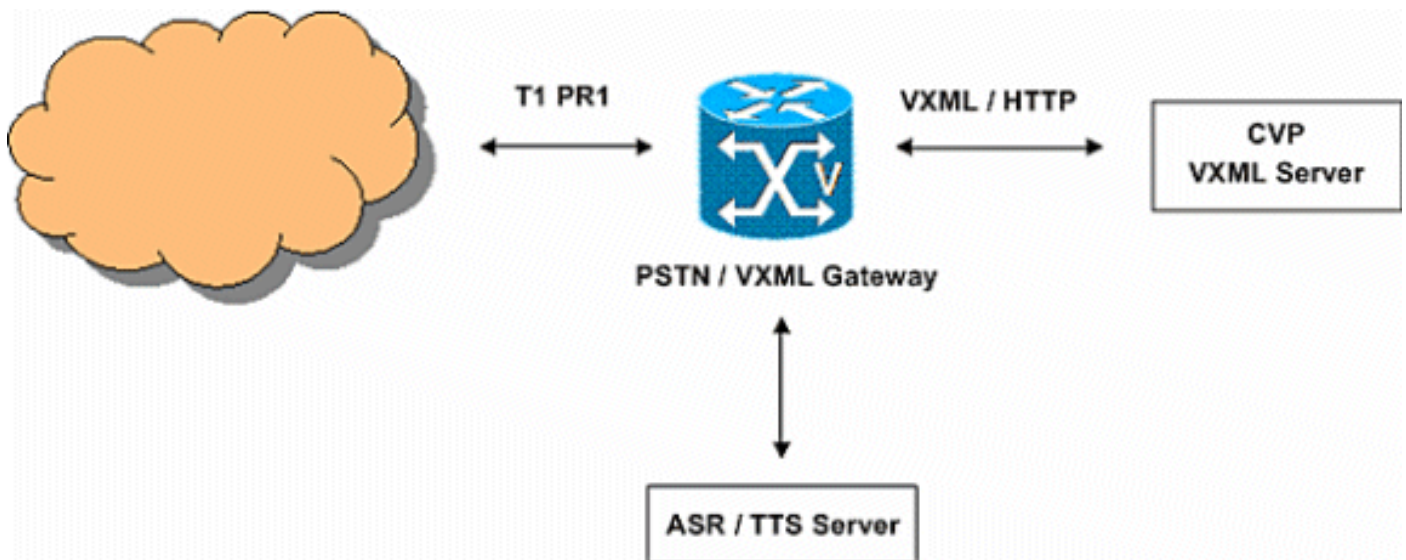
Configurar

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

Note: 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 !--- CVPPPrimary parameter specifies
the !--- IP address of the VXML server service Pharmacy
flash:CVPSelfService.tcl paramspace english index 0
paramspace english language en paramspace english
location flash: param CVPSelfService-port 7000 param
CVPSelfService-app GoodPrescriptionRefillApp7 paramspace
english prefix en param CVPPPrimaryVXMLServer
172.18.110.75 !--- Specifies the Gateway's RTP !---
stream to the ASR / TTS to go around the !--- Content
Service Switch !--- instead of through the CSS. mrcp
client rtpsetup enable !--- Specify the maximum memory
size !--- for the HTTP Client Cache http client cache
memory pool 15000 !--- Specify the maximum number of
file !--- that can be stored in the !--- HTTP Client
Cache http client cache memory file 500 !--- Disable
Persistent !--- HTTP Connections no http client

```

```

connection persistent !--- Configure the T1 PRI
controller T1 3/0 framing esf linecode b8zs pri-group
timeslots 1-24 !--- Configure the ISDN switch !---- type
and incoming-voice !---- under the D-channel interface
interface Serial3/0:23 no ip address encapsulation hdlc
isdn switch-type primary-net5 isdn incoming-voice modem
no cdp enable ! --- Configure a POTS !---- dial-peer
that will be used !---- as inbound dial-peer for calls
coming ! --- in across the T1 PRI line. !---- The
"pharmacy"service !---- is applied under this dial-peer.
dial-peer voice 1 pots service pharmacy destination-
pattern 5555 direct-inward-dial port 3/0:D forward-
digits all !--- Configure a SIP Voip !---- dial-peer
that will be used !---- as an outbound dial-peer when
the !---Gateway initiates a MRCP overc SIP !---- session
to the ASR server. !---- Codec = G711ulaw, DTMF-Relay !-
--- = RTP-NTE, No Vad dial-peer voice 5 voip session
protocol sipv2 destination uri ASR dtmf-relay rtp-nte
codec g711ulaw no vad !--- Configure a SIP Voip !----
dial-peer that will be used !---- as an outbound dial-
peer when the !---Gateway initiates a MRCP !---- overc
SIP session to the TTS server !--- Codec = G711ulaw,
DTMF-Relay = RTP-NTE, !---- No Vad dial-peer voice 6
voip session protocol sipv2 destination uri TTS dtmf-
relay rtp-nte codec g711ulaw no vad

```

Exemplo do 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 documento/página VXML.
6. O Gateway de IOS executa o documento VXML.
7. Se o documento 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 documento 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 documento 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. Depois que a execução do documento VXML, o gateway envia um pedido do CARGO HTTP (com um conjunto de parâmetro) como especificado na etiqueta do <submit> do documento/página VXML.
11. As etapas 6 – 10 ocorrem para cada documento VXML enviado pelo server.
12. Quando o aplicativo VXML termina o serviço proporcionado ao chamador, envia um documento 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 active voice brief**

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

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

- **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/allF8 :
164 333360890ms.1 +20 pid:5 Originate
sip:asr@172.18.110.76:5060 active

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

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

• 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”.

Note: Consulte [Informações Importantes sobre Comandos de Depuração](#) antes de usar comandos **debug**.

Note: 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**
- **cancela 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 mrp**
- **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 entregue 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 documento VXML \(1\). Este documento 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 documento VXML \(1\).](#)
11. [O gateway recebe a APROVAÇÃO 200 para seu pedido do HTTP do CARGO. O corpo da mensagem contém o documento VXML \(2\). Este documento 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 documento VXML \(2\).](#)
13. [O gateway recebe uma resposta de 200 APROVAÇÕES para o pedido do CARGO HTTP. O corpo da mensagem contém o documento VXML \(3\). Este documento 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 DTMF 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-COMLETE" 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 documento 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 documento 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 documento 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 entregue 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 documento VXML (1). O documento 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&
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//HTTTPC:/httpc_socket_read:
*Jan 18 03:34:55.647:
  read data from the socket 3
  : first 400 bytes of data:
HTTP/1.1 200 OK
Content-Length: 26450
Content-Type: audio/wav
Last-Modified:
  Mon, 30 Apr 2007 15:36:51 GMT
Accept-Ranges: bytes
ETag: "e0c1445f3d8bc71:2d6"
Server: Microsoft-IIS/6.0
Date: Mon, 30 Apr 2007 16:58:42 GMT
Connection: close
```

```
RIFFJg(Unprintable char...)
  0057415645666D7420120001010401
  F00401F00108000666163744000176700
  64617461176700FFFFFFF807
  FFFFFFFF80FFFFFFF80F
(other hex information not shown).
```

[O gateway envia um pedido do HTTP do CARGO ao server como definido no "submete" a opção 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 uma APROVAÇÃO 200 para seu pedido do HTTP do CARGO](#)

O corpo da mensagem contém o documento VXML (2). O documento 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&
  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 documento VXML \(2\)](#)

```

*Jan 18 03:34:55.667:
  //127//HTTTPC:/httpc_write_stream:
  Client write buffer fd(4):
POST /CVP/Server HTTP/1.1
Host: 172.18.110.75:7000
Content-Length: 67
Content-Type:
  application/x-www-form-urlencoded
Cookie: $Version=0; JSESSIONID=
  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 documento VXML (3). Este documento 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&
  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" xmlns="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/xxxxxxxxxxxxx/CCAPI/ccCallSetupRequest:

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

*Jan 18 03:34:57.527:
// -1/xxxxxxxxxxxxx/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/xxxxxxxxxxxxx/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/xxxxxxxxxxxxx/CCAPI/

ccIFCallSetupRequestPrivate:

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

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

Called Number=sip: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/xxxxxxxxxxxxx/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/xxxxxxxxxxxxx/CCAPI

/ccIFCallSetupRequestPrivate:

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

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

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

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

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

[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/xxxxxxxxxxxxx/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/xxxxxxxxxxxxx/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=mrpcv2_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/xxxxxxxxxxxxx/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"?>
  <say version="1.0" xml:lang="en-us">
    Thank you for calling Audium pharmacy.</say>
```

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-COMLETE" depois que falou a alerta obrigado

MRCP/2.0 141 SPEAK-COMLETE 1 COMPLETE

Channel-Identifier:
000023EC46361276@speechsynth

Completion-Cause: 000 normal

Speech-Marker: ""

O chamador de PSTN incorpora "1" para escolher o preenchimento

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 documento 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-urlencoded

ed; len=258; iov_base=audium_vxmlLog=%7C%7C%7Caudio_
_group$$$initial_audio_group
%5E%5E%5E4%7C%7C%7Cutterance
$$$1%5E%5E%5E15340%7C%7C
%7Cinputmode$$$dtmf%5E%5E%5E15344%
7C%7C%7Cinterpretation$$$refills
%5E%5E%5E15344%7C%7C%7Cconfidence$$$0
%5E%5E%5E15344&confidence=0&
choice_fld=refills

*Jan 18 03:35:12.931:
about to send data to the socket 3
: first 400 bytes of data:

POST /CVP/Server HTTP/1.1

Host: 172.18.110.75:7000

Content-Length: 258

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

Cookie: \$Version=0; JSESSIONID=
BBCE0F948ADFDB720497F587A7997538;
\$Path=/CVP

Connection: close

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

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

[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 documento 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>

<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(Responded=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](#)
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