Media Resource Control Protocol v2
A Tutorial

Sarvi Shanmugham,

Editor: MRCP v1/v2

Technical Leader, Cisco Systems

Roadmap

- Overview of the IETF Speechsc WG Effort
- MRCP – Short Summary
- MRCP –Architecture Diagram
- MRCP - Usage
- MRCP v1 & v2 – Current Status

Overview of the IETF Speechsc WG Effort

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- IETF Working group - formed in 2002
- Aimed to develop a protocol that allows distributed speech processing(speech recognition, speaker recognition, verification and text-to-speech)
- Work with VoiceXML and SALT
- Leverage existing protocols as much as possible
- Leverage existing W3C standards for markup

MRCP – Short Summary (contd.)

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- **Basic Speech Services defined**

  Speech Recognition

  Text-to-Speech

  Speaker Identification

  Speaker Verification

  Recording

MRCP – The Framework

- **The MRCP Framework leverages a suite of protocols and XML markup to achieve its purposes and only fills in where the needs have not already been addressed.**

  SIP – This is used for discovering MRCP resources in the network and to rendezvous with the server and establish the necessary control and media pipes to the resources.

  SDP – SDP is used in conjunction with SIP for both resource discovery and the setup of control and media pipes for the session.

  RTP/RTCP – This is used for media transmission to/from the media processing resources.

  MRCP – This controls the operation of individual media processing resources, like ASR, TTS, SI, SV and recorders.

MRCP – The Framework (contd.)

- **W3C markup specifications**

  SRGS – Definition of Voice Grammars that are processed by Speech Recognition engines.

  N-Grams – Stochastic Grammars.

  Semantic Tags – The above grammars could contain semantic markup associated with the grammars that aids in semantic processing of the recognized texts.

  SSML – Definitions Speech markup to be processed by Text-To-Speech Engines.

  NLSML – Natural Language Semantic Markup Language

MRCP – The Framework (contd.)

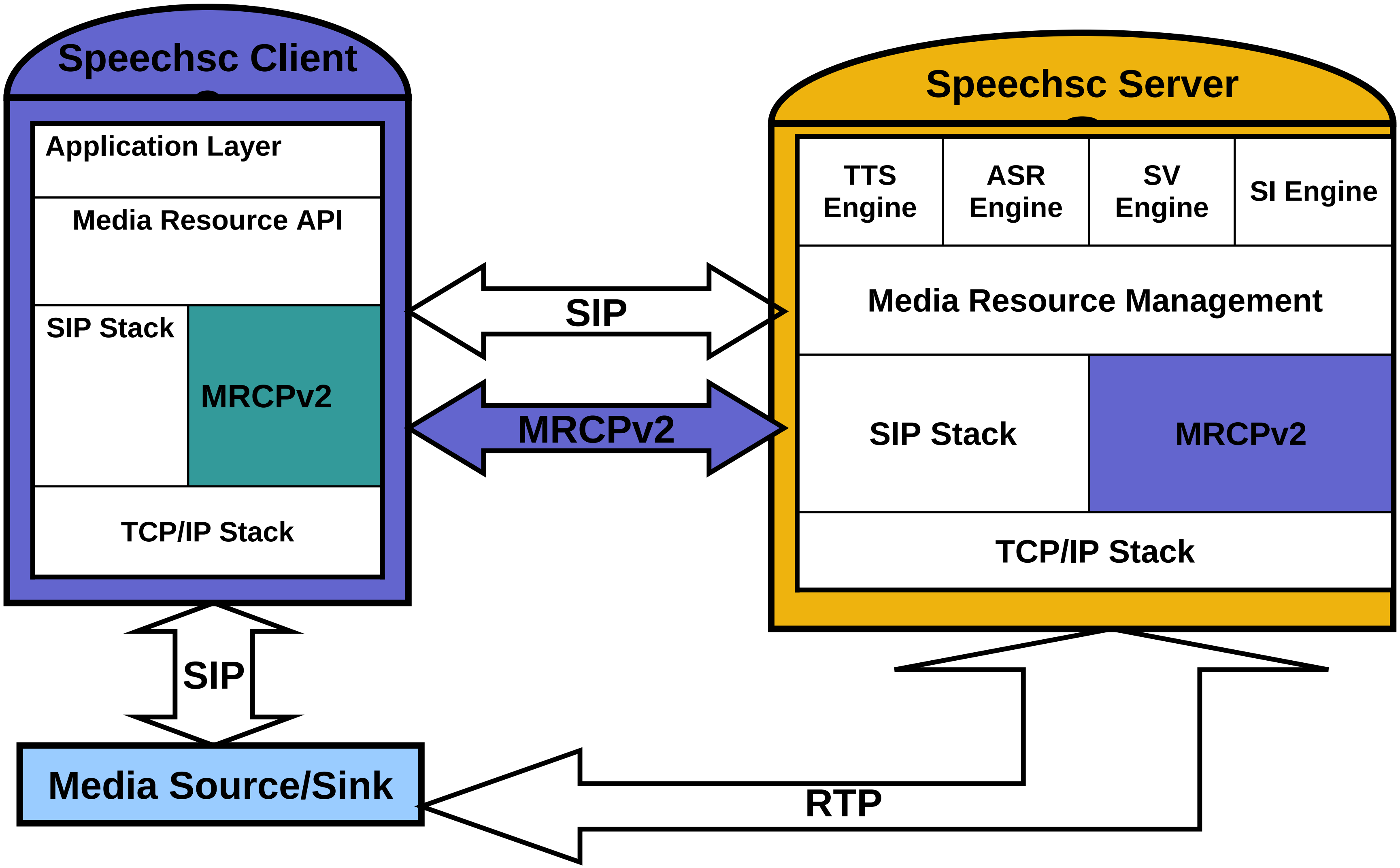
- **MRCP enhancements**

  Recognition Results – The recognition resource returns results as a markup that is primarily based on NLSML. But there are a few minor additions to fill in gaps not addressed by NLML

  Grammar Enrollment Results – When enrolling new grammars, the results XML returned also contains extra information describing the enrollment status of the grammar enrollment.

  Speaker Identification/Verification Results – When doing Speaker Verification or Identification these XML extensions allow the resource to return the results of the verification or identification operation.

MRCP – Architecure Diagram



Server and Resource Addressing

- **Server**

  It’s a regular SIP URI like the one below

  sip:mrcpv2@mediaserver.com

- **Resource Addressing**

  speechrecog - Speech Recognition

  dtmfrecog - DTMF Recognition

  speechsynth - Speech Synthesis

  basicsynth - Poorman's Speech Synthesizer

  speakverify - Speaker Verification

  recorder - Speech Recording

MRCPv2 Protocol Basics

- **Connecting to the Server**

  Uses a SIP INVITE and the SDP offer/answer model to connect to the media server and establish the session media and control pipes.

  Uses m= audio …. For setting up media pipes to the server. This is the same as in any other SIP call setup.

  The m-line media stream established can shared by multiple mrcpv2 resource that may be part of the same SIP session.

  Uses m=control …. For setting up individual control pipes for each MRCPv2 resource that the client wants to control.

  There is one m=control .. line in the offer for every resource the client wants to allocate for the session.

  The m-lines specifies a transport type of TCP, SCTP or TLS and a fromat type of application/mrcpv2. The port number of this line MUST contain 9(discard port) in the offer and a valid server port in the answer. The client may then initiate an appropriate transport connection that port.

MRCPv2 Protocol Basics

- **Connecting to the Server**

  The offer m-line from the client also contains an “resource” specifying what type of resource it wants to allocate for the session. The corresponding answer m-line must contain a “channel” attribute that contains a channel identifier that will be used in all MRCP messages between the client and that specific resource.

  The transport connection(TCP, SCTP or TLS) could be shared across multiple MRCP sessions between a client and server.

- **Channel-Idenitifier**

  A channel identifier allocated for each resource is of the form

  32AECB234338@speechsynth

- **De-Allocating a Resource**

  To de-allocate a resource the client issues a SIP re-INVITE to the server where the appropriate m=control …. lines port is 0.

MRCPv2 Protocol Basics

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```
INVITE sip:mresources@mediaserver.com SIP/2.0
Via: SIP/2.0/TCP client.atlanta.example.com:5060;
      branch=z9hG4bK74bf9
Max-Forwards: 6
To: MediaServer <sip:mresources@mediaserver.com>
From: sarvi <sip:sarvi@cisco.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314161 INVITE
Contact: <sip:sarvi@cisco.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=sarvi 2890844526 2890842808 IN IP4 126.16.64.4
s=-
c=IN IP4 224.2.17.12
m=control 9 TCP application/mrcpv2
a=resource:speechsynth
a=cmid:1
m=audio 49170 RTP/AVP 0 96
a=rtpmap:0 pcmu/8000
a=recvonly
a=mid:1
```

MRCPv2 Protocol Basics


```
SIP/2.0 200 OK
Via: SIP/2.0/TCP client.atlanta.example.com:5060;
        branch=z9hG4bK74bf9
To: MediaServer <sip:mresources@mediaserver.com>
From: sarvi <sip:sarvi@cisco.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314161 INVITE
Contact: <sip:sarvi@cisco.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=sarvi 2890844526 2890842808 IN IP4 126.16.64.4
s=-
c=IN IP4 224.2.17.12
m=control 32416 TCP application/mrcpv2
a=channel:32AECB234338@speechsynth
a=cmid:1
m=audio 48260 RTP/AVP 00 96
a=rtpmap:0 pcmu/8000
a=sendonly
a=mid:1
```

MRCPv2 Protocol Basics

ACK sip:mresources@mediaserver.com SIP/2.0

Via: SIP/2.0/TCP client.atlanta.example.com:5060;

branch=z9hG4bK74bf9 Max-Forwards: 6

To: MediaServer <sip:mresources@mediaserver.com>;tag=a6c85cf

From: Sarvi <sip:sarvi@cisco.com>;tag=1928301774

Call-ID: a84b4c76e66710

CSeq: 314162 ACK

Content-Length: 0

Types of MRCP Messages

- **Request**

  MRCP/2.0 434 SPEAK 543260

  Channel-Identifier: 32AECB23433802@speechsynth

  Voice-gender: neutral

  ………

- **Response**

  MRCP/2.0 48 543260 200 IN-PROGRESS

  Channel-Identifier: 32AECB23433802@speechsynth

  ………

- **Event**

  MRCP/2.0 73 SPEAK-COMPLETE 543260 COMPLETE

  Channel-Identifier: 32AECB23433802@speechsynth

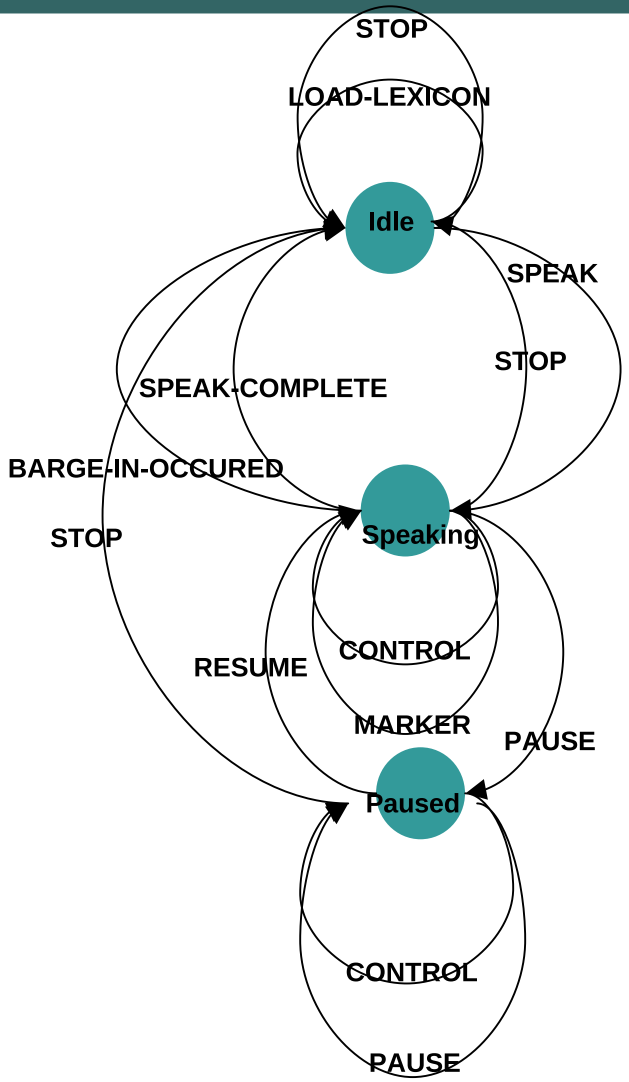
  ………

Generic Messages

- **Request**
  - **SET-PARAMS**
  - **GET-PARAMS**
- **Headers**
  - **Channel-Identifier**
  - **Active-Request-Id-List**
  - **Proxy-Sync-Id**
  - **Content-Id**
  - **Content-Type**
  - **Content-Length**
  - **Content-Base**
  - **Content-Location**
  - **Content-Encoding**
  - **Cache-Control**
  - **Logging-Tag**
  - **Set-Cookie**
  - **Set-Cookie2**
  - **Vendor-Specific**

Text-To-Speech Resource

- **Request**
  - SPEAK
  - STOP
  - PAUSE
  - RESUME
  - BARGE-IN-OCCURRED
  - CONTROL
  - LOAD-LEXICON
- **Event**
  - SPEECH-MARKER
  - SPEAK-COMPLETE



Text-To-Speech Resource

- **Headers**

  - Jump-Target
  - Kill-On-Barge-In
  - Speaker-Profile
  - Completion-Cause
  - Completion-Reason
  - Voice-Parameter
  - Prosody-Parameter
  - Speech-Marker
  - Speech-Language
  - Fetch-hint
  - Audio-Fetch-Hint
  - Fetch-Timeout
  - Failed-Uri
  - Failed-uri-cause
  - Speak-Restart
  - Speak-Length
  - Load-Lexicon
  - Lexicon-Search-Order

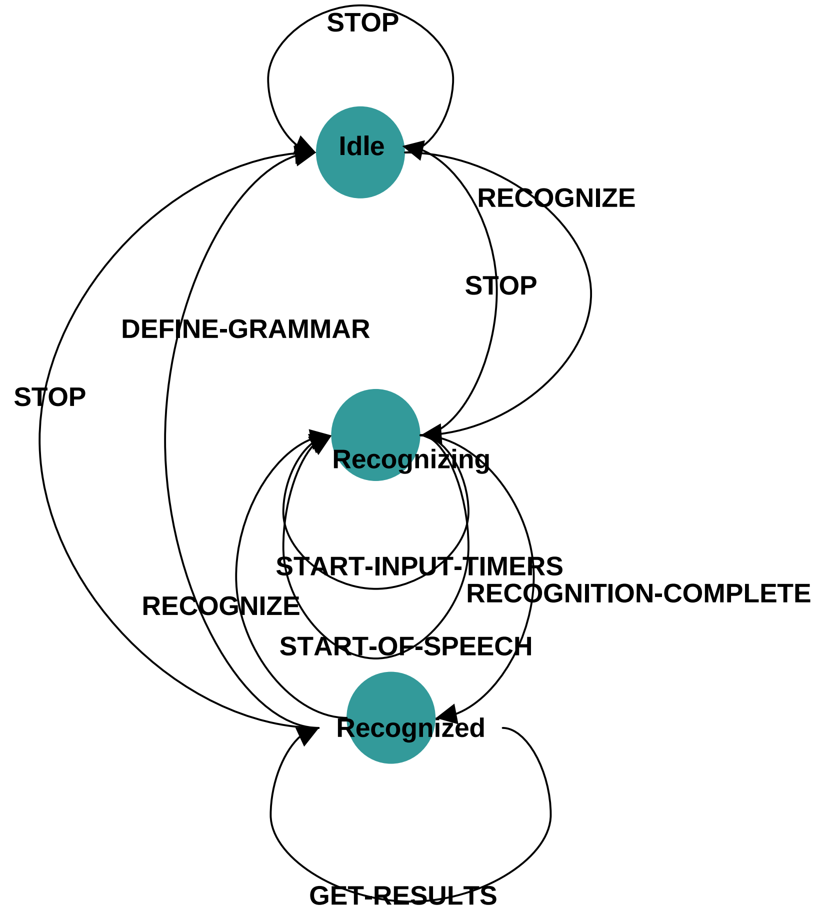
Text-To-Speech Resource

Speech Markup

```
<?xml version="1.0"?>
<speak>
<paragraph>
    <sentence> You have 4 new messages. </sentence>
    <sentence>The first is from <say-as type="name"> Stephanie
              Williams </say-as> and arrived at <break/>
              <say-as type="time"> 3:45pm </say-as>.
    </sentence>
    <sentence>The subject is <prosody rate="-20%"> ski
              trip </prosody>
    </sentence>
</paragraph>
</speak>
```

Recognition Resource

- **Request**
  - DEFINE-GRAMMAR
  - RECOGNIZE
  - INTERPRET
  - GET-RESULT
  - START-INPUT-TIMERS
  - STOP
  - START-PHRASE-ENROLLMENT
  - ENROLLMENT-ROLLBACK
  - END-PHRASE-ENROLLMENT
  - MODIFY-PHRASE
  - DELETE-PHRASE
- **Event**
  - START-OF-SPEECH
  - RECOGNITION-COMPLETE
  - INTERPRETATION-COMPLETE



Recognition Resource

- **Recognition Headers**
  - Confidence-Threshold
  - Sensitivity-Level
  - Speed-Vs-Accuracy
  - N-Best-List-Length
  - No-Input-Timeout
  - Recognition-Timeout
  - Waveform-Url
  - Completion-Cause
  - Completion-Reason
  - Recognizer-Context-Block
  - Start-Input-Timers
  - Speech-Complete-Timeout
  - Speech-Incomplete-Timeout
  - Dtmf-Interdigit-Timeout
  - Dtmf-Term-Timeout
  - Dtmf-Term-Char
  - Fetch-Timeout
  - Failed-Uri
  - Failed-Uri-Cause
  - Save-Waveform
  - New-Audio-Channel
  - Speech-Language
  - Ver-Buffer-Utterance
  - Recognition-Mode
  - Cancel-If-Queue
  - Hotword-Max-Duration
  - Hotword-Min-Duration
  - Interpret-text

Recognition Resource

- **Enrollment Headers**

  Num-Min-Consistent-Pronunciations

  Consistency-Threshold

  Clash-threshold

  Personal-Grammar-Uri

  Phrase-Id

  Phrase-NL

  Weight

  Save-Best-Waveform

  New-Phrase-Id

  Confusable-Phrases-Uri

  Abort-Phrase-Enrollment

Recognition Resource

Grammar Markup

```
<?xml version="1.0"?>
<!-- the default grammar language is US
   English -->
<grammar xml:lang="en-US" version="1.0">
<!-- single language attachment to
   tokens -->
<rule id="yes">
            <one-of>
                <item xml:lang="fr-
   CA">oui</item>
                <item xml:lang="en-
   US">yes</item>
            </one-of>
   </rule>
<!-- single language attachment to a
   rule expansion -->
   <rule id="request">
            may I speak to
            <one-of xml:lang="fr-CA">
                <item>Michel
   Tremblay</item>
                <item>Andre Roy</item>
            </one-of>
   </rule>
```

```
<!-- multiple language attachment to a
   token -->
   <rule id="people1">
            <token lexicon="en-US,fr-
   CA"> Robert </token>
   </rule>
   <!-- the equivalent single-language
   attachment
                 expansion -->
   <rule id="people2">
            <one-of>
                <item xml:lang="en-
   US">Robert</item>
                <item xml:lang="fr-
   CA">Robert</item>
            </one-of>
   </rule>
   </grammar>
```

Recognition Resource

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Result Markup

```
<?xml version="1.0"?>
<result
 grammar="http://theYesNoGrammar">
    <interpretation>
        <instance>
            <myApp:yes_no>

  <response>yes</response>
            </myApp:yes_no>
        </instance>
        <input>ok</input>
    </interpretation>
</result>
```

Recognition Resource

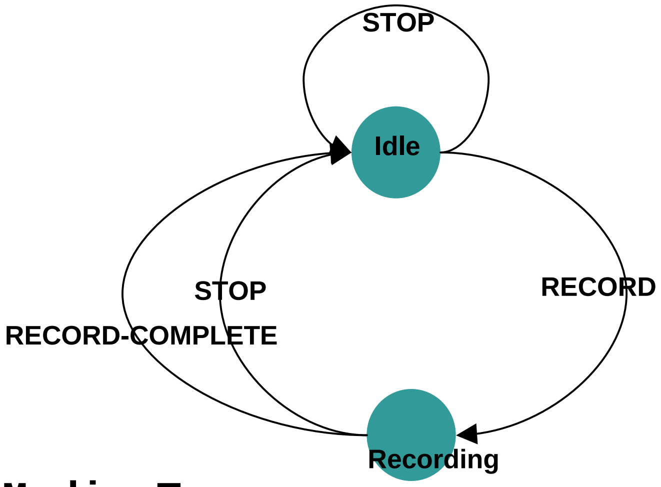
Enrollment Result Markup

```
<?xml version= "1.0"?>
<result grammar="Personal-Grammar-URI“
  xmlns:mrcp=
        "http://www.ietf.org/mrcp2">
<mrcp:result-type type="ENROLLMENT"/>
 <mrcp:enrollment-result>
     <num-clashes> 2 </num-clashes>
     <num-good-repetitions> 1
        </num-good-repetitions>
     <num-repetitions-still-needed> 1
        </num-repetitions-still-needed>
     <consistency-status> consistent
        </consistency-status>
     <clash-phrase-ids>
  <item> Jeff </item>
  <item> Andre </item>
</clash-phrase-ids>
```

```
<transcriptions>
    <item> m ay b r ow k er </item>
    <item> m ax r aa k ah </item>
   </transcriptions>
   <confusable-phrases>
     <item>
       <phrase> call </phrase>
       <confusion-level> 10
          </confusion-level>
     </item>
   </confusable-phrases>
</mrcp:enrollment-result>
</result>
```

Recording Resource

- Request
  - RECORD
  - STOP
  - START-INPUT-TIMERS
- Event
  - START-OF-SPEECH
  - RECORD-COMPLETE
- Headers



`Sensitivity-Level`	`Media-Type`
`No-Input-Timeout`	`Max-Time`
`Completion-Cause`	`Final-Silence`
`Completion-Reason`	`Capture-On-Speech`
`Failed-Uri`	`Ver-Buffer-Utterance`
`Failed-Uri-Cause`	`Start-input-timers`
`Record-Uri`	`New-audio-channel`

Verification Resource

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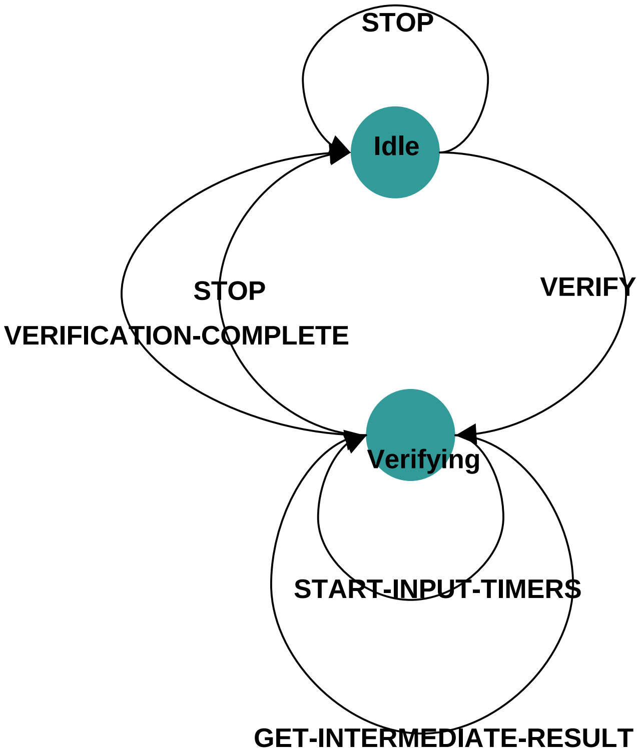
- **Request**
  - **START-SESSION**
  - **END-SESSION**
  - **QUERY-VOICEPRINT**
  - **DELETE-VOICEPRINT**
  - **VERIFY**
  - **VERIFY-FROM-BUFFER**
  - **VERIFY-ROLLBACK**
  - **STOP**
  - **CLEAR-BUFFER**
  - **START-INPUT-TIMERS**
  - **GET-INTERMEDIATE-RESULT**
- **Event**
  - **VERIFICATION-COMPLETE**
  - **START-OF-SPEECH**



Verification Resource

- **Verification Headers**

  `Repository-Uri`

  `Voiceprint-Identifier`

  `Verification-Mode`

  `Adapt-Model`

  `Abort-Model`

  `Security-Level`

  `Num-Min-Verification-Phrases`

  `Num-Max-Verification-Phrases`

  `No-Input-Timeout`

  `Save-Waveform`

  `Waveform-Url`

  `Voiceprint-Exists`

  `Ver-Buffer-Utterance`

  `Input-Waveform-Url`

  `Verification-Type`

  `Digit-Sequence`

  `Completion-Cause`

  `Completion-Reason`

  `Speech-Complete-Timeout`

  `New-Audio-Channel`

  `Abort-Verification`

  Start-Input-Timers

Verification Resource

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Verification Result Markup

```
<?xml version="1.0"?>
<result grammar="What-Grammar-URI"
  xmlns:mrcp="http://www.ietf.org/mrcp2">
  <mrcp:result-type type="VERIFICATION" />
  <mrcp:verification-result>
    <voiceprint id="johnsmith">
      <adapted> true </adapted>
      <incremental>
        <num-frames> 50 </num-frames>
        <device> cellular-phone </device>
        <gender> female </gender>
        <decision> accepted </decision>
        <verification-score> 0.98514 </verification-score>
      </incremental>
      <cumulative>
        <num-frames> 1000 </num-frames>
        <device> cellular-phone </device>
```

Verification Resource

Verification Result Markup(contd.)

```
        <gender> female </gender>
        <decision> accepted </decision>
        <verification-score> 0.91725</verification-score>
      </cumulative>
    </voiceprint>
    <voiceprint id="marysmith">
      <cumulative>
        <verification-score> 0.93410 </verification-score>
      </cumulative>
    </voiceprint>
    <voiceprint uri="juniorsmith">
      <cumulative>
        <verification-score> 0.74209 </verification-score>
      </cumulative>
    </voiceprint>
  </mrcp:verification-result>
</result>
```

Call Flow Example

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```
C->S:
INVITE sip:mresources@mediaserver.com SIP/2.0
Max-Forwards: 6
To: MediaServer <sip:mresources@mediaserver.com>
From: sarvi <sip:sarvi@cisco.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314163 INVITE
Contact: <sip: sarvi@cisco.com>
Content-Type: application/sdp
Content-Length: 142

v=0
o=sarvi 2890844526 2890842809 IN IP4 126.16.64.4
s=SDP Seminar
i=A session for processing media
c=IN IP4 224.2.17.12/127
m=control 9 SCTP application/mrcpv2
a=resource:speechsynth
a=cmid:1
m=audio 49170 RTP/AVP 0 96
a=rtpmap:0 pcmu/8000
a=recvonly
a=mid:1
```

```
m=control 9 SCTP application/mrcpv2
a=resource:speechrecog
a=cmid:2
m=audio 49180 RTP/AVP 0 96
a=rtpmap:0 pcmu/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=sendonly
a=mid:2
```

Call Flow Example


```
S->C:
SIP/2.0 200 OK
To: MediaServer <sip:mresources@mediaserver.com>
From: sarvi <sip:sarvi@cisco.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314163 INVITE
Contact: <sip: sarvi@cisco.com>
Content-Type: application/sdp
Content-Length: 131

v=0
o=sarvi 2890844526 2890842809 IN IP4 126.16.64.4
s=SDP Seminar
i=A session for processing media
c=IN IP4 224.2.17.12/127
m=control 32416 SCTP application/mrcpv2
a=channel:32AECB23433801@speechsynth
a=cmid:1
m=audio 48260 RTP/AVP 0
a=rtpmap:0 pcmu/8000
a=sendonly
a=mid:1
m=control 32416 SCTP application/mrcpv2
a=channel:32AECB23433802@speechrecog
a=cmid:2
m=audio 48260 RTP/AVP 0
a=rtpmap:0 pcmu/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=recvonly
a=mid:2

C->S:
ACK sip:mrcp@mediaserver.com SIP/2.0
Max-Forwards: 6
To: MediaServer
      <sip:mrcp@mediaserver.com>;tag=a6c85cf
From: Sarvi <sip:sarvi@cisco.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314164 ACK
Content-Length: 0
```

Call Flow Example

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```
C->S: MRCP/2.0 386 SPEAK 543257
Channel-Identifier: 32AECB23433802@speechsynth
Kill-On-Barge-In: false
Voice-gender: neutral
Voice-category: teenager
           Prosody-volume: medium
Content-Type: application/synthesis+ssml
Content-Length: 104

<?xml version="1.0"?>
<speak>
<paragraph>
         <sentence>You have 4 new
      messages.</sentence>
         <sentence>The first is from <say-as
         type="name">Stephanie Williams</say-as>
      <mark name="Stephanie"/>
         and arrived at <break/>
         <say-as type="time">3:45pm</say-
      as>.</sentence>

         <sentence>The subject is <prosody
         rate="-20%">ski
      trip</prosody></sentence>
</paragraph>
</speak>
```

```
S->C: MRCP/2.0 49 543257 200 IN-PROGRESS
Channel-Identifier: 32AECB23433802@speechsynth

S->C: MRCP/2.0 46 SPEECH-MARKER 543257 IN-
      PROGRESS
Channel-Identifier: 32AECB23433802@speechsynth
           Speech-Marker: Stephanie
```

The synthesizer finishes with the SPEAK request.

```
S->C: MRCP/2.0 48 SPEAK-COMPLETE 543257 COMPLETE
Channel-Identifier: 32AECB23433802@speechsynth
```

Call Flow Example

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```
C->S:MRCP/2.0 343 RECOGNIZE 543258
Channel-Identifier: 32AECB23433801@speechrecog
Content-Type: application/grammar+xml
Content-Length: 104

<?xml version="1.0"?>

<!-- the default grammar language is US English
      -->
<grammar xml:lang="en-US" version="1.0">

<!-- single language attachment to a rule
      expansion -->
      <rule id="request">
          Can I speak to
          <one-of xml:lang="fr-CA">
                   <item>Michel Tremblay</item>
                   <item>Andre Roy</item>
          </one-of>
      </rule>

</grammar>

S->C: MRCP/2.0 49 543258 200 IN-PROGRESS
Channel-Identifier: 32AECB23433801@speechrecog
```

```
C->S: MRCP/2.0 289 SPEAK 543259
Channel-Identifier: 32AECB23433802@speechsynth
           Kill-On-Barge-In: true
Content-Type: application/sml
Content-Length: 104

<?xml version="1.0"?>
<speak>
<paragraph>
         <sentence>Welcome to ABC
      corporation.</sentence>
         <sentence>Who would you like Talk
      to.</sentence>
</paragraph>
</speak>

S->C: MRCP/2.0 52 543259 200 IN-PROGRESS
Channel-Identifier: 32AECB23433802@speechsynth
```

Call Flow Example

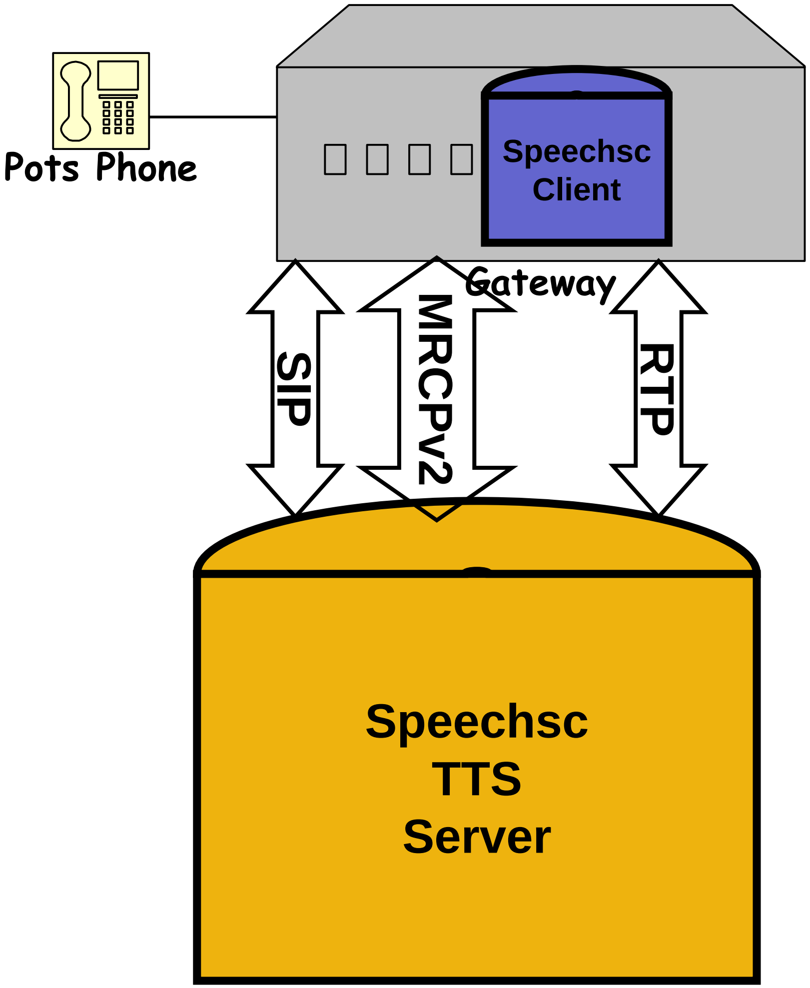

```
S->C: MRCP/2.0 49 START-OF-SPEECH 543258 IN-PROGRESS
Channel-Identifier: 32AECB23433801@speechrecog
          Proxy-Sync-Id: 987654321


C->S: MRCP/2.0 69 BARGE-IN-OCCURRED 543259
Channel-Identifier: 32AECB23433802@speechsynth
          Proxy-Sync-Id: 987654321

S->C: MRCP/2.0 72 543259 200 COMPLETE
Channel-Identifier: 32AECB23433802@speechsynth
          Active-Request-Id-List: 543258

S->C: MRCP/2.0 73 SPEAK-COMPLETE 543259 COMPLETE
Channel-Identifier: 32AECB23433802@speechsynth
          Completion-Cause: 001 barge-in

S->C: MRCP/2.0 412 RECOGNITION-COMPLETE 543258 COMPLETE
Channel-Identifier: 32AECB23433801@speechrecog
          Completion-Cause: 000 success
Waveform-URL: http://web.media.com/session123/audio.wav
Content-Type: application/x-nlsml
Content-Length: 104
```

```
<?xml version="1.0"?>
<result x-model="http://IdentityModel"
  xmlns:xf="http://www.w3.org/2000/xforms"
  grammar="session:request1@form-level.store">
    <interpretation>
        <xf:instance name="Person">
            <Person>
                <Name> Andre Roy </Name>
            </Person>
        </xf:instance>
        <input>   may I speak to Andre Roy </input>
    </interpretation>
</result>
```

```
C->S:BYE sip:mrcp@mediaserver.com SIP/2.0
Max-Forwards: 6
From: Sarvi <sip:sarvi@cisco.com>;tag=a6c85cf
To: MediaServer
      <sip:mrcp@mediaserver.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0
```

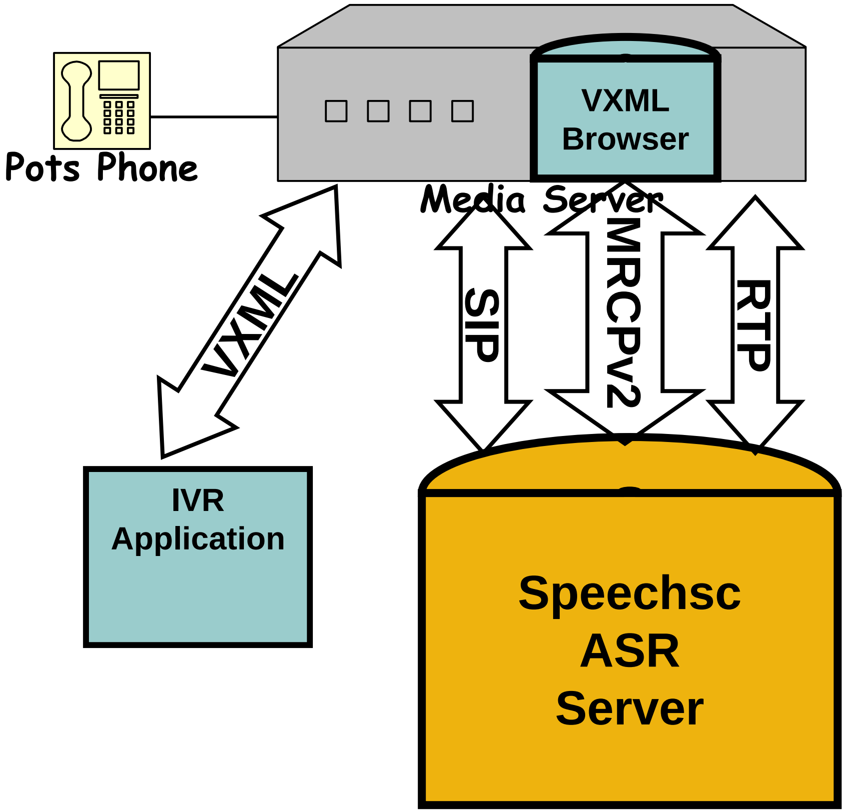
Use Case: Text to Speech Announcements

- **POTS phone attempts call.**
- **VoIP gateway, acting as a SIP UA, attempts SIP session to complete the call; gets error, like "486 Busy Here".**
- **VoIP Gateway constructs a text error string from the SIP message, such as "Your call to 978-555-1212 did not go through because the called party was busy".**
- **Gateway INVITES SPEECHSC server to connect RTP stream and issues an MRCPv2 TTS request for the error message**
- **Speechsc server plays message to the user on the POTS phone.**



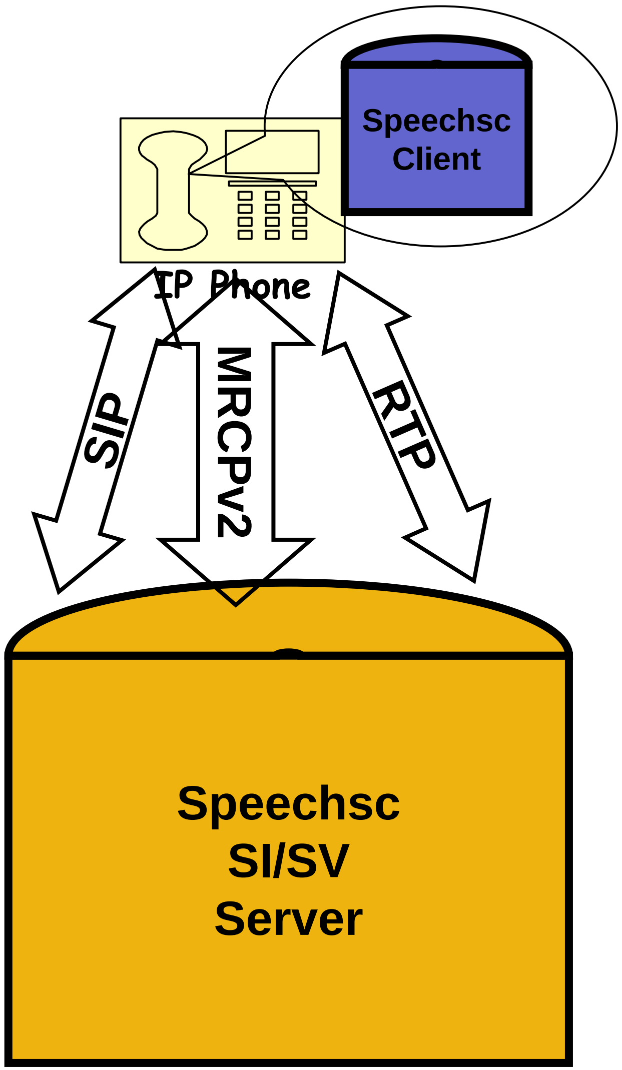
Use Case: VXML-based ASR

- Users call into the service in order to obtain stock quotes.
- Media Server fetches VoiceXML to drive user interaction.
- Media Server INVITEs Speechsc server for ASR
- VoiceXML interpreter on the Media Server directs the user's media stream to the ASR server and uses MRCPv2 to control the ASR server.
- Results come back and the application proceeds.



Use Case: Speaker Verification

- A user speaks into a SIP phone to "log in" to that phone to make and receive phone calls using his identity and preferences
- IP phone uses SIP and MRCPv2 to set up an RTP stream between the phone and the SPEECHSC SI/SV server and request verification.
- SV server verifies the user's identity and returns the result via MRCPv2.
- The IP Phone may either use the identity directly to identify the user in outgoing calls, to fetch the user's preferences from a configuration server, request authorization from a AAA server, etc.



Current WG Status

- **Requirements Document passed IESG Review - soon to be published as an RFC**

  draft-ietf-speechsc-reqts-05.txt

- **MRCPv2 Protocol Document in second revision - expect last call in late fall**

  draft-ietf-speechsc-mrcpv2-04.txt

- **MRCPv1 Protocol Document is pending IESG review for publication as an Informational RFC.**

  http://www.ietf.org/internet-drafts/draft-shanmugham-mrcp-05.txt