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№1 в России по созданию ИТ-инфраструктур

ЧТО ТАКОЕ VOICEXML И CCXML?

- VoiceXML
  - Language for interacting with a person via telephone
  - Can speak to person via recorded prompts or TTS
  - Can get input from person via DTMF or ASR
- CCXML
  - Language for managing telephone calls
  - Can initiate/terminate/conference calls
  - Once call is established, call is typically handled by a VoiceXML application

VOICE PORTAL ATTRIBUTES

- Runs on industry standard hardware
  - Intel x86
- Runs on industry standard operating system
  - Red Hat ES6.0
- Uses industry standard protocols to communicate with external systems
  - HTTPS
  - MRCP
  - SIP

VOICE PORTAL OFFERS

- Bundled solution includes:
  - Hardware
  - Linux

    Note – The Linux provided by Avaya is a (small) subset of Red Hat ES6 Update 3.
  - Voice Portal
  - Dialog Designer
- Software-only solution includes:
  - Voice Portal
  - Dialog Designer

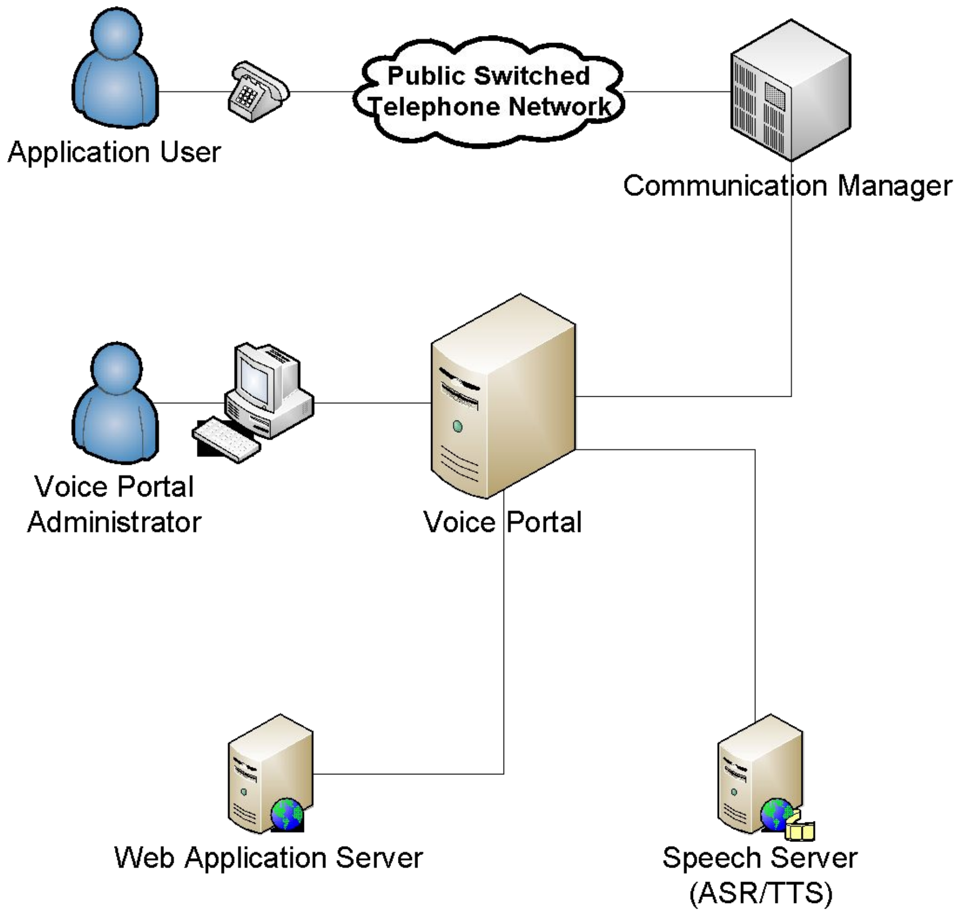
VOICE PORTAL СОВМЕСТИМОСТЬ

- PBX
  - Communication Manager (CM) 2.1+
- SIP gateway
  - Avaya Session Management (SM) 6.0+
- Web application server
  - Dialog Designer applications require Apache Tomcat, BEA WebLogic, IBM WebSphere, or IBM WebSphere Express
  - Non-Dialog Designer applications can run on any application server

VOICE PORTAL СОВМЕСТИМОСТЬ

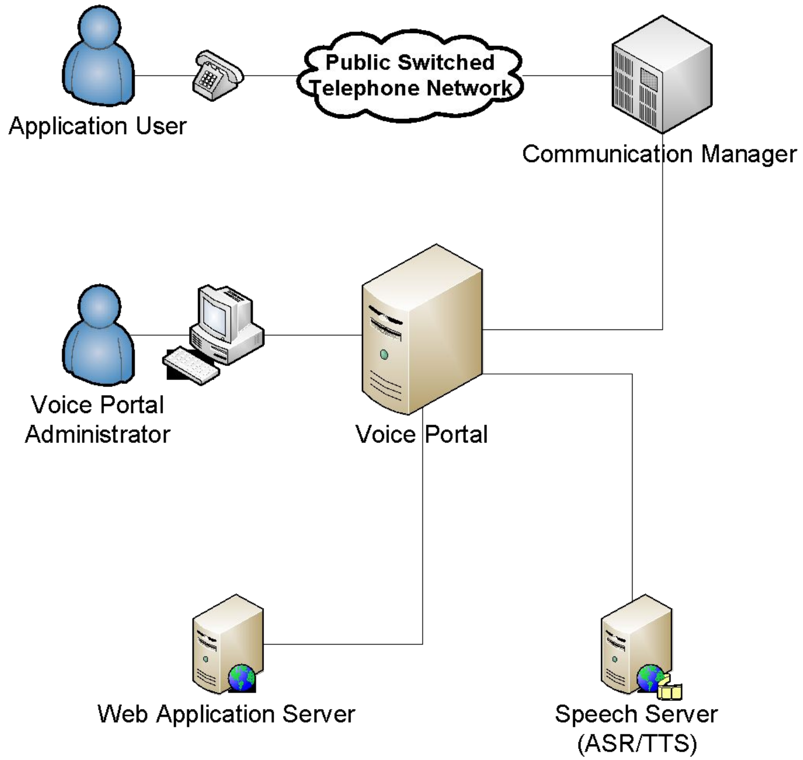
- Speech servers
  - IBM WebSphere Voice Server 5.1.3+ (ASR+TTS)
  - Nuance RealSpeak 4.0.12+ (TTS)
  - Nuance Recognizer 9.0+ (ASR)
  - Nuance OpenSpeech Recognizer 3.0.13+ (ASR)

VP ОБЩАЯ СХЕМА



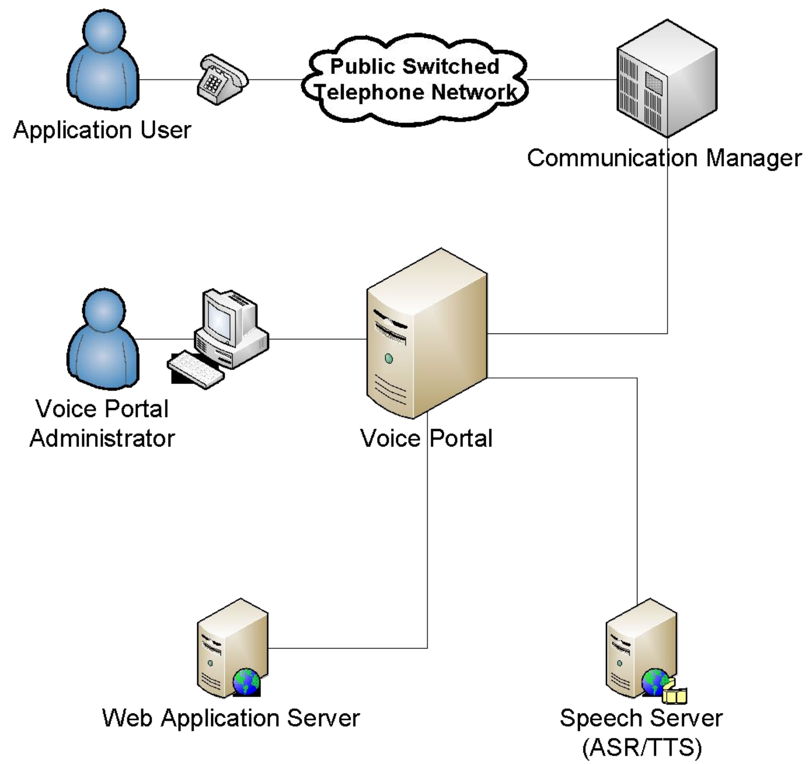
SIMPLIFIED CALL FLOW

1. Caller dials a phone number that routes through PSTN to CM.
2. CM routes call to Voice Portal.
3. Based on number dialed by caller, Voice Portal accesses appropriate application URL on web server.
4. Web server launches speech application, which is typically written in a high level programming language such as Java.



SIMPLIFIED CALL FLOW (CONT.)

5. Speech application running on web server generates VoiceXML that gets returned to Voice Portal.
6. VoiceXML gets processed by Voice Portal, which causes Voice Portal to interact with caller.
7. When VoiceXML requests ASR or TTS, Voice Portal contacts appropriate speech server.
8. When caller hangs up, application is terminated.



VOICE PORTAL COMPONENTS

- Voice Portal Management System (VPMS)
  - Provides web interface for configuring system.
  - Houses Voice Portal database that stores all configuration data.
  - Distributes telephony ports among MPPs.
  - Provides outcall web service.
- Media Processing Platform (MPP)
  - Talks to PBX to initiate/answer calls.
  - Contains CCXML and VoiceXML interpreters that process applications.
  - Talks to speech servers to provide ASR/TTS support.

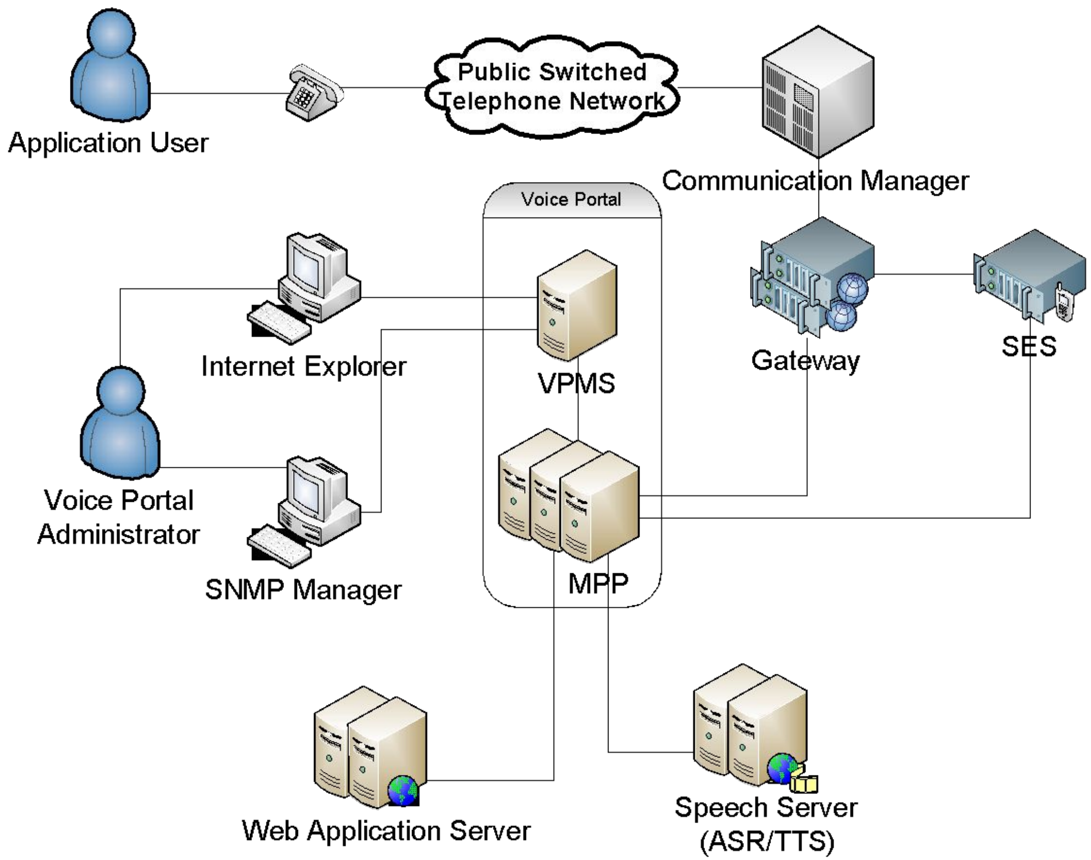
VOICE PORTAL CONFIGURATIONS

- Single-box
  - VPMS and MPP run on one server.

  Note – If single server goes down, entire system is out of service.
- Multi-box
  - Single VPMS on a dedicated server.
  - Up to 20 MPPs on separate servers.

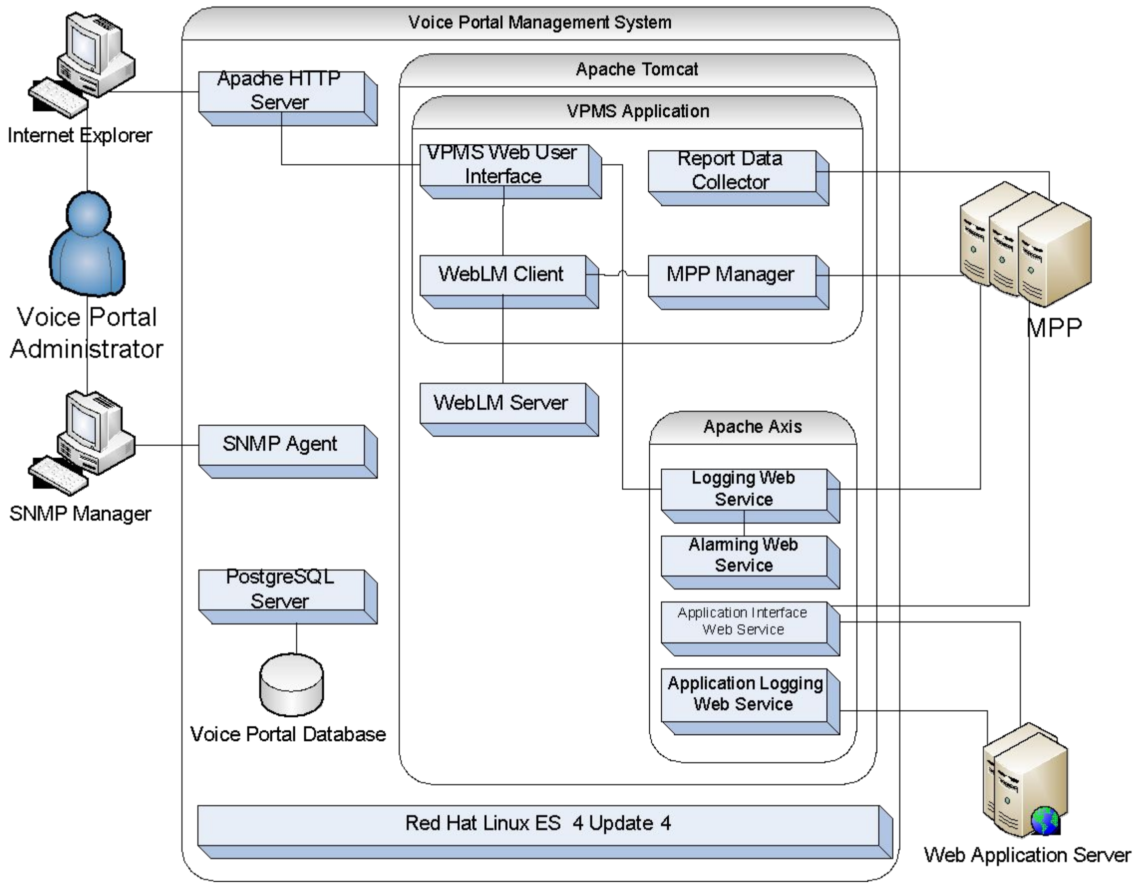
  Note – If single server goes down, surviving servers are still able to process calls.

АРХИТЕКТУРА VOICE PORTAL



Important – In order to ensure adequate network performance, VPMS, MPPs, gateways, and speech servers should all be physically co-located.

VPMS ARCHITECTURE



Note – Virtually all VPMS components go through PostgreSQL Server to access Voice Portal Database. The lines were omitted to enhance readability.

VPMS COMPONENTS

- VPMS Application
  - Web user interface
    - Browser interface administrator uses to manage Voice Portal system.
    - Allows administrator to configure system, start/stop system and get status, and generate reports.
  - MPP manager
    - Controls MPPs.
    - Sends configuration parameters and commands entered by administrator through web user interface to MPPs.
    - Monitors status of MPPs and redistributes telephony resources as necessary.

VPMS COMPONENTS (CONT.)

- VPMS Application (cont.)
  - Report data collector
    - Collects call detail and session detail report data from MPPs.
  - WebLM client
    - Collects license information from license server.
- SNMP agent
  - Interface that allows administrator to query Voice Portal status using third party Simple Network Management Protocol (SNMP) manager.

VPMS COMPONENTS (CONT.)

- Web services
  - Logging web service
    - Logs events that are displayed using VPMS Log Viewer.
  - Alarming web service
    - Logs events that are displayed using VPMS Alarm Manager.
    - Generates SNMP notifications.
  - Application Logging web service
    - Logs events that are displayed using VPMS report Application Detail.
  - Application Interface web service
    - Initiates an outcall and/or launches a speech application.

VPMS COMPONENTS (CONT.)

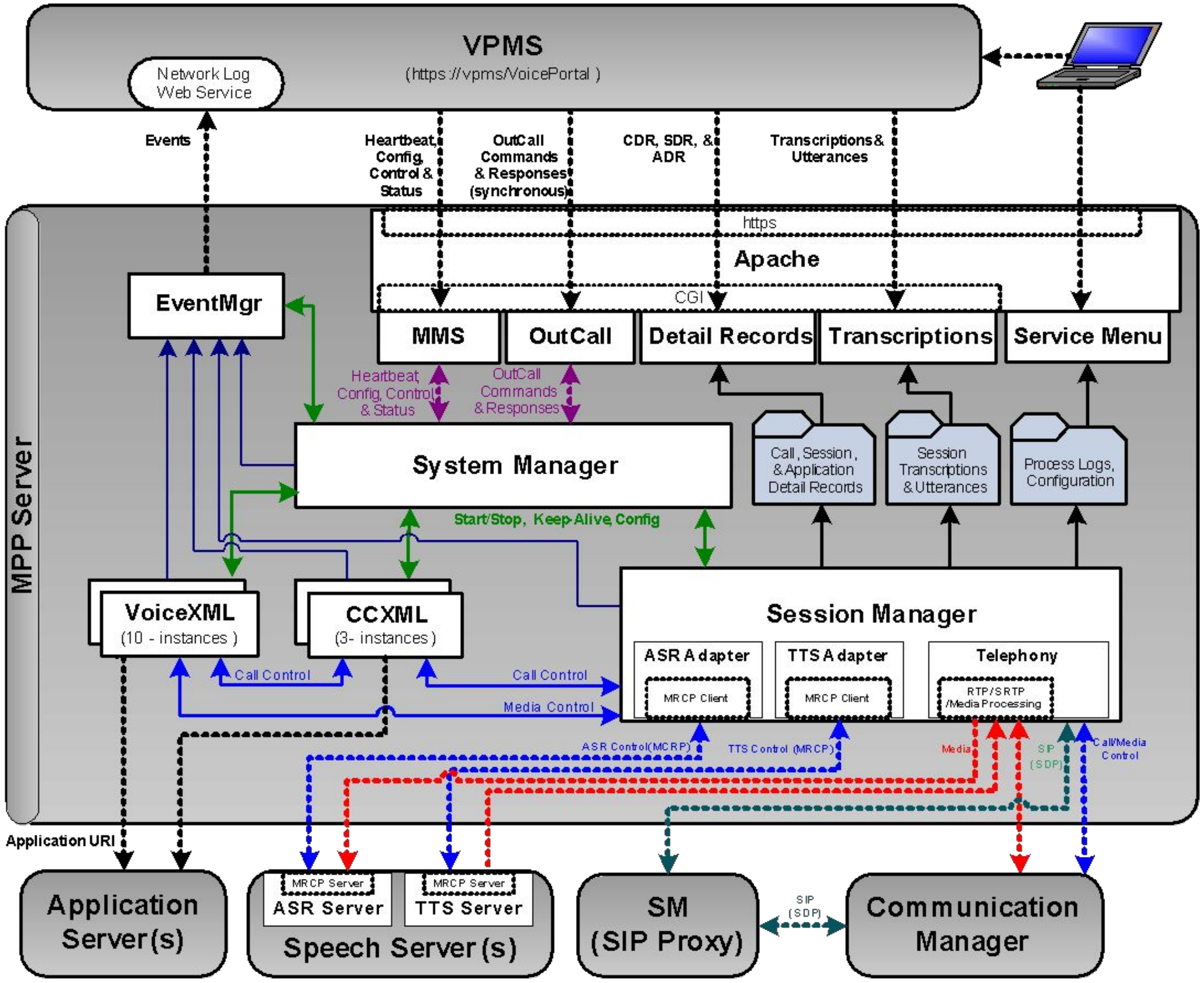
- WebLM server
  - License server
- Voice Portal database
  - Contains Voice Portal configuration information.
  - Contains log/report data.

Important – This is the thing that you need to back-up!

MPP – FEATURES

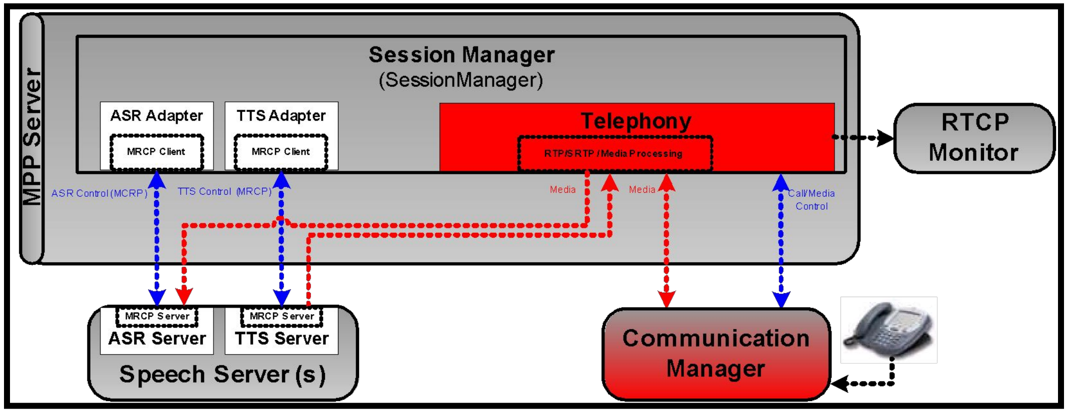
- Runs on Linux (RH ES 6 Update 2 or greater)
- 100% software implementation
- VoiceXML 2.1 and CCXML 1.0 applications
- MRCP 1/2 (ASR/TTS)
- Telephony – H.323 & SIP
- No local administration or configuration
- No persistent data (no need for MPP backups)
- Automatic restart of crashed/hung processes
- Logging – process, call/session, application, & performance

MPP ARCHITECTURE – PROCESS VIEW

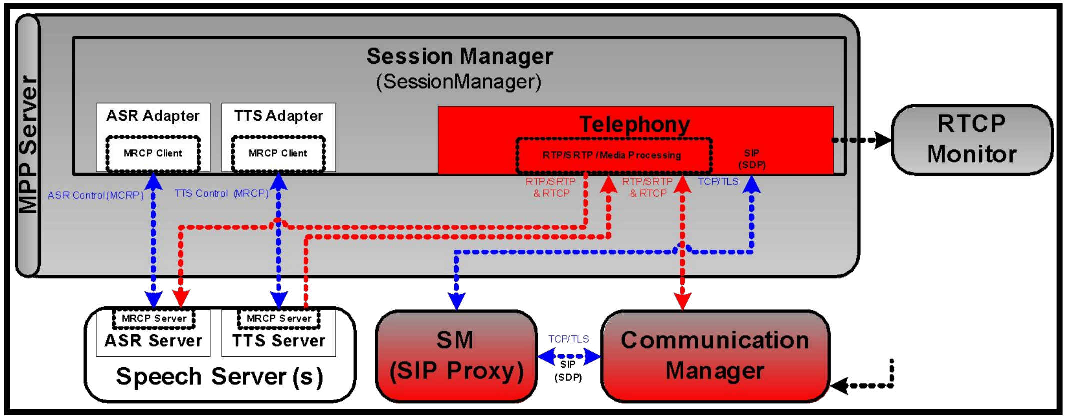


MPP –TELEPHONY

H323



SIP



Feature	H.323	H.323 + Green Feature (CM 3.1 build 369 or later)	SIP
URI types	*tel:*	*tel:*	*tel:*, *sip:*, *sips:*
Application Selection	Dialed number Ex: 7777	Dialed number Ex: 8002227777	*To* or *Request URI* (wildcards via regex) Ex: bob@avaya.com, 7777@avaya.com
Blind Transfer	✓	✓	✓
Bridge Transfer	✓ Note: no call status (e.g. busy)	✓	✓ Note: *connecttimeout* does not work.
Supervised/Consultative Transfer	Works like Blind Transfer Note: Do get a "noroute" status	✓	✓
DTMF	**detection via CM**	**detection via CM**	**detection via RFC2833**
Play (e.g. prompt.wav, prompt.vis)	✓	✓	✓
Record	✓	✓	✓
Vectoring Converse-on step	✓	✓	
UUI			✓ **plus SIP Headers!**
Encryption	• Disabled • AES • AEA	• Disabled • AES • AEA	• Disabled • TLS • SRTP
QOS	✓	✓	✓
Failover	Configure alternate GateKeeper in VPMS	Configure alternate GateKeeper in VPMS	SES performs load balancing/failover/Busy

MPP – TELEPHONY – URI TYPES

- Used by VP during:
  - Inbound Call:
    - Incoming Call’s Originating & Destination/Called number
  - Outbound Call:
    - VXML `<transfer>` tag
    - Argument in the Outcall Web Service
- 3 types
  - *tel:*
  - *sip:*
  - *sips:*

MPP – TELEPHONY – URI TYPES: ‘*TEL:*’

- What you are used to using from VP 3.0
- H323 & SIP
- Supported chars: 0123456789*#
- Supported PostDial Digits: ***,pP*** plus ***0123456789*#ABCD***
- Example:

```
<transfer name=“transfer” dest=“tel:1234567,888”>
```

MPP – TELEPHONY – URI TYPES: *'SIP:'* & *'SIPS:'*

- **sip:**
  - SIP Only
  - URI Format (See [RFC3261](RFC3261) (See RFC3261 (Section 19.1.1), [RFC2396](RFC2396) (See RFC3261 (Section 19.1.1), RFC2396, http://www.iana.org/assignments/sip-parameters for more detail):

    ```
    sip:user:password@host:port;uri-parameters?headers
    ```

    Examples:

    ```
    <transfer name=“transfer” dest=“sip:bob@avaya.com”>
    <transfer name=“transfer”
      dest=“sip:alice@atlanta.com;maddr=239.255.255.1;ttl=15”>
    <transfer name=“transfer”
      dest=“sip:alice@avaya.com?subject=project&priority=urgent”>
    ```
- **sips:**
  - SIP Only
  - ***sips:*** is *Secure SIP*.
  - Uses same URI format as ***sip:***
  - Be sure to configure your CM, SES, and VPMS to use TLS.
    Warning! Call will fail if it can't use secure, encrypted transport (TLS).
  - Example:

    ```
    <transfer name=“transfer”
      dest=“sips:alice@atlanta.com;transport=tcp”>
    ```

MIXING H.323 & SIP – OUTBOUND PORT SELECTION

- Dependent on the URI type

  tel:

  1. H.323 outbound port available? If no, go to step 2.
  2. SIP outbound channel available? If no, go to step 3. If yes, convert ***tel:*** URI to a ***sip:*** For example,
     `tel:8037332555` becomes `sip:8037332555@avaya.com`
     where `avaya.com` is the **Route Domain** configured in the VPMS's **SIP Connection.**
  3. Call fails with a NO RESOURCE (`error.connection.noresource` thrown in VXML).

  sip:

  1. SIP outbound (TCP or TLS) channel available. If no, go to step 2.
  2. Call fails with a NO RESOURCE (`error.connection.noresource` thrown in VXML).

  sips:

  1. SIP outbound (TLS) channel available. If no, go to step 2.
  2. Call fails with a NO RESOURCE (`error.connection.noresource` thrown in VXML).

- Bridged Transfers adds a twist
  1. Outbound port available from the ***same*** port group/trunk used by the incoming call? If no, go to step 2.
  2. Use algorithm defined in previous bullet to find a port/channel.

MPP – TELEPHONY – SIP HEADERS & UUI

- SIP Headers
  - "`Name: Value`" data included in the SIP Requests/Responses
  - CCXML & VXML apps can get and set headers (*more on this later*)
  - VP provides access to limited SIP Headers:

    Call-ID, *Contact*, *From*, *To*, *History-Info*, *P-asserted Identity*, *Require*, *Supported*, *User-to-User*, *User-Agent*, *Via*, *User-To-User Information (UUI)*, & *Unknown*
- UUI (User-to-User Information)
  - VoiceXML & CCXML call it ***AAI*** (Application-to-Application Information)
  - Is a SIP Header
    - *Inbound*: both CM (`User-to-User`) and AudioCodes (`X-UserToUser`) format are checked
    - *Outbound*: UUI passed in the SIP Header, `User-to-User` or app can send customized name using `Unknown` header
  - Example: Passing collected data (account number) in Call Center applications

MPP – TELEPHONY - SIP HEADERS &

VXML page for Bridge Transfer → SIP INVITE sent from the MPP

```
<?xml version="1.0" encoding="UTF-8"?>
<vxml version="2.0"
   xmlns="http://www.w3.org/2001/vxml">
    <form id="xfer">
         <var name="mydur" expr="0"/>
         <block>
              <prompt>Transfer</prompt>
         </block>
         <transfer
            name="mycall"
            dest="tel:92312782"
            aai="Put your UUI here"
            type="bridge"
            transferaudio="mohal.wav"
            connecttimeout="10s">
         </transfer>
    </form>
</vxml>
```

```
INVITE sip:92312782@columbia.avaya.com;user=phone SIP/2.0
From:
      <sip:2707@columbia.avaya.com;user=phone>;tag=669bbe9c356dc1
      200094877585
To: <sip:92312782@columbia.avaya.com;user=phone>
Call-ID: a29bbe9c356dc1300094877585
CSeq: 1 INVITE
Max-Forwards: 70
Route: <sip:135.148.133.6;transport=tcp;lr>
Via: SIP/2.0/TCP
      135.148.133.117;branch=z9hG4bKaa2be9c356dc1400094877585
User-Agent: AVP UA 4.0
Supported: 100rel, timer, histinfo
Allow: INVITE, CANCEL, BYE, ACK, PRACK, NOTIFY, OPTIONS
Contact: <sip:135.148.133.117;transport=tcp>
Session-Expires: 1200;refresher=uac
Min-SE: 180
Content-Type: application/sdp
User-to-User:Put your UUI here
Content-Length: 159

v=0
o=- 1 1 IN IP4 135.148.133.117
s=-
c=IN IP4 135.148.133.117
t=0 0
m=audio 30006 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
```

MPP – TELEPHONY – QUALITY OF SERVICE (QOS)

- Setting Prioritization
  - Supported in VP for both SIP & H.323
  - Used by intervening routers to prioritize packet throughput
  - Supported on layers 2 and 3 of the RTP stack
  - Configurable from VPMS on ***System Configuration*** > ***MPP Servers*** > ***VoIP Settings*** > ***QoS Parameters***
  - To work all equipment along the RTP path must support QOS
- Monitoring
  - Set RTCP Monitor Settings on VPMS
  - MPP’s Telephony provider sends statistics during active session

MPP – ASR & TTS

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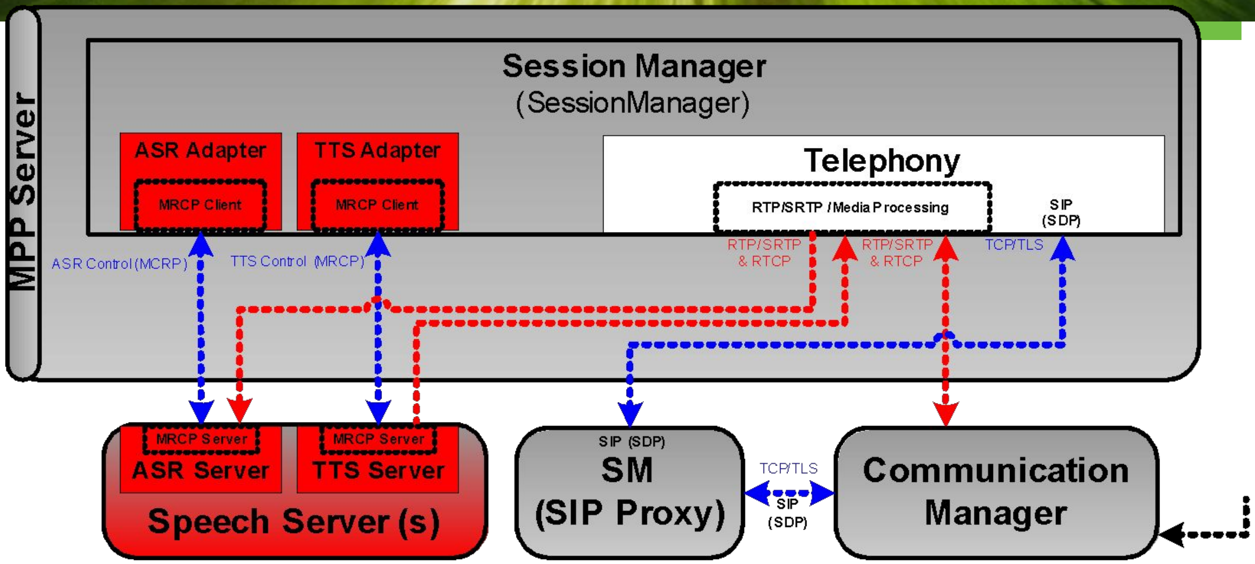

- MPP uses MRCP (v.1 draft 4) to talk to Speech Servers
- Supported Speech Servers: IBM WVS, Nuance, Nuance Quantum
- For multilingual applications, we recommend the Speech Servers in your Voice Portal configuration be installed with all languages needed by the application

MPP – ASR & TTS - SPEECH SERVER COMPARISON

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Feature	IBM	Nuance	Nuance Quantum
Minimum Versions (recommended)	Websphere Voice Server (WVS) R5.1.3 (R5.1.3 + Fix Pack 3)	**ASR**: OSR 3.0.10 (3.0.13) **TTS**: Realspeak 4.0.10 (4.0.12) **SWMS**: 3.1.10 (3.1.14)	**ASR**: Recognizer 9 **TTS**: Realspeak 4.5 **Speech Server**: 5.0
OS Supported	• Linux • Windows	• Linux • Windows	• Linux • Windows
Load Balancing	• Voice Portal implementation • IBM Load Balancer	• Voice Portal implementation	• Voice Portal implementation
Languages (TTS)	• 8 Languages & Dialects • >13 voices	• 24 Languages & Dialects • >30 voices	• 26 Languages & Dialects • >35 voices
Languages (ASR)	• 8 Languages	• 46 Languages	• 44 Languages
Speech Recognition Grammar Specification (SRGS)	**Version 1.0**	**Version 1.0**	**Version 1.0**
Speech Synthesis Markup Language (SSML)	**Version 1.0**	**Version 1.0**	**Version 1.0**
N-best, Hot Word, Barge-in, Custom Dictionary	✓	✓	✓
OSDM & OSD		OSDM 2.0.4	OSDM 2.1
DTMF Processing	✓	✓	✓

MPP – ASR & TTS – RESOURCE ALLOCATION

- On the VPMS each ASR/TTS server is configured with *Total Number of Licensed ASR/TTS Resources*

ASR | TTS

☐	Name	Network Address	Engine Type	Base Port	Total Number of Licensed TTS Resources	Voices
☐	IBM TTS	135.148.133.95	IBM WVS	554	7	English(US) en-US Andrew M, English(US) en-US Allison F
☐	Nuance Realspeak	135.148.133.99	Nuance RealSpeak	4900	5	English(US) en-US Jennifer F

Add Delete Help

- For Each ASR/TTS Server MPP dynamically allocates ports based on
  - *ASR/TTS Server's Total Number of Licensed ASR/TTS Resources*
  - Total telephony ports/channels for VP System
  - Number of telephony ports/channels for the MPP
- MPP automatically adjusts allocation when ASR/TTS Server configuration or Telephony Ports/Channel count changes.
  - If increase in speech ports, takes effect immediately
  - If decrease in speech ports, may be delayed if ports are in use

MPP – ASR & TTS – RESOURCE LOAD BALANCING

- Load Balancing is ***not*** across all MPPs in a VP system.
  It is local to an MPP.
- Speech Servers can have different license capacities
- The MPP determines which ASR/TTS server to use by looking at:
  - Speech Server state (up/down)
  - Language(s) required by the application
  - Speech Server status (errors, & latencies)
  - Speech Server with the least in-use ports (on that MPP)

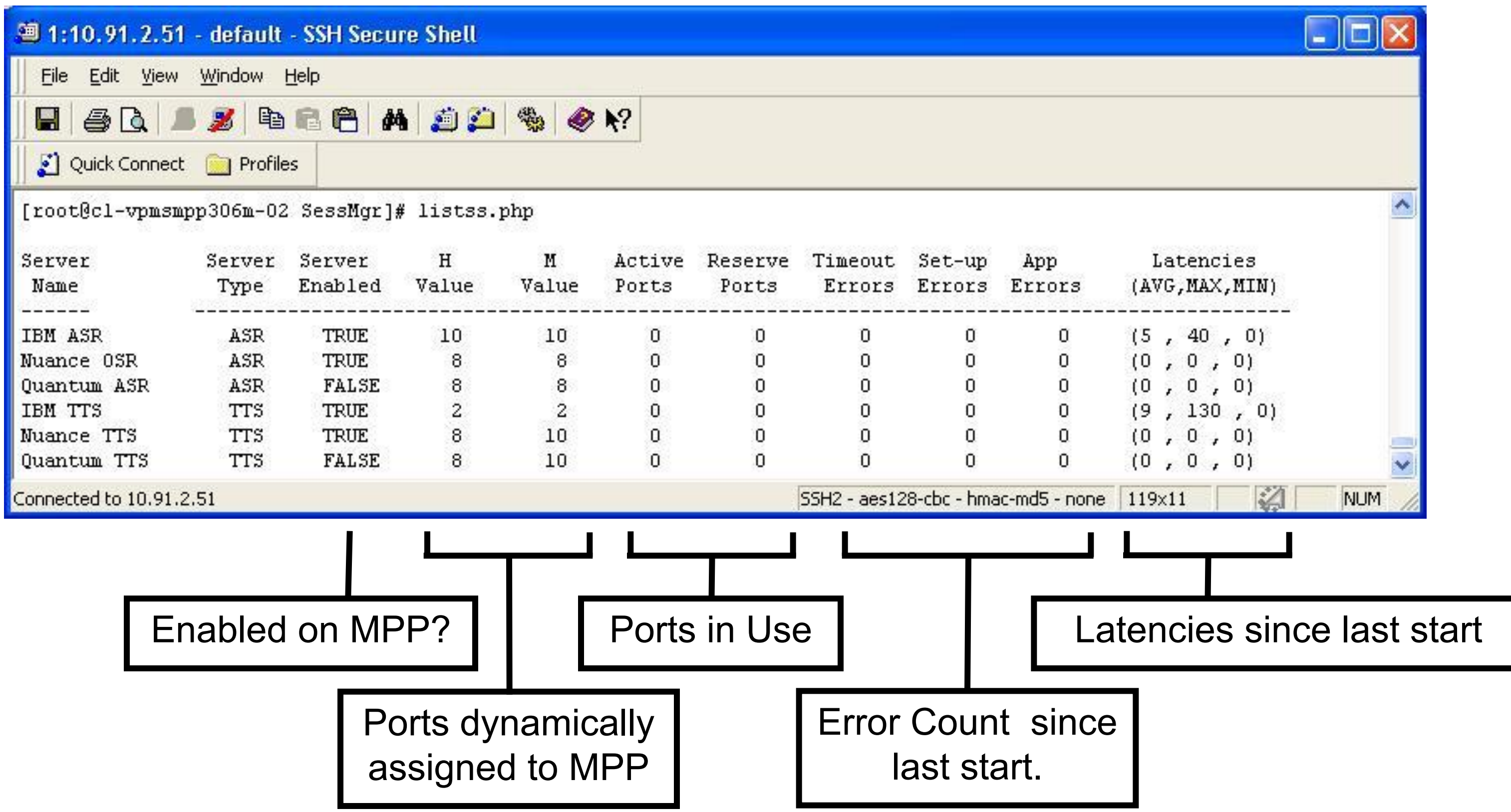
MPP – ASR & TTS – RESOURCE MULTIPLEXING

- Multiplexing saves you $$$, through purchasing fewer Speech Server ports.
- TTS Resources are multiplexed
  - The same TTS resource can be used by multiple simultaneous calls.

    A TTS resource is allocated to a call when a TTS prompt is queued and returned to a free pool when the play is complete (the resource hasn't been released). This allows for another call, requiring the same resource, to utilize the already established connection.
  - If your application uses minimal TTS (more prerecorded prompts), then you could potentially purchase fewer TTS licenses.
- ASR Resources are ***not*** multiplexed. No $$$ saved here…
  - *Total # ASR licenses* = *Total # Telephony ports* across the VP System
  - An ASR port is allocated at the start of a session and released at the end of a session.

MPP – ASR & TTS – MONITORING

- Use MPP Service Menu or *listss.php* on MPP to view the status for the Speech Servers and their allocated resources




```
[root@cl-vpmsmpp306m-02 SessMgr]# listss.php
```

Server Name	Server Type	Server Enabled	H Value	M Value	Active Ports	Reserve Ports	Timeout Errors	Set-up Errors	App Errors	Latencies (AVG,MAX,MIN)
IBM ASR	ASR	TRUE	10	10	0	0	0	0	0	(5 , 40 , 0)
Nuance OSR	ASR	TRUE	8	8	0	0	0	0	0	(0 , 0 , 0)
Quantum ASR	ASR	FALSE	8	8	0	0	0	0	0	(0 , 0 , 0)
IBM TTS	TTS	TRUE	2	2	0	0	0	0	0	(9 , 130 , 0)
Nuance TTS	TTS	TRUE	8	10	0	0	0	0	0	(0 , 0 , 0)
Quantum TTS	TTS	FALSE	8	10	0	0	0	0	0	(0 , 0 , 0)

MPP – APPLICATIONS

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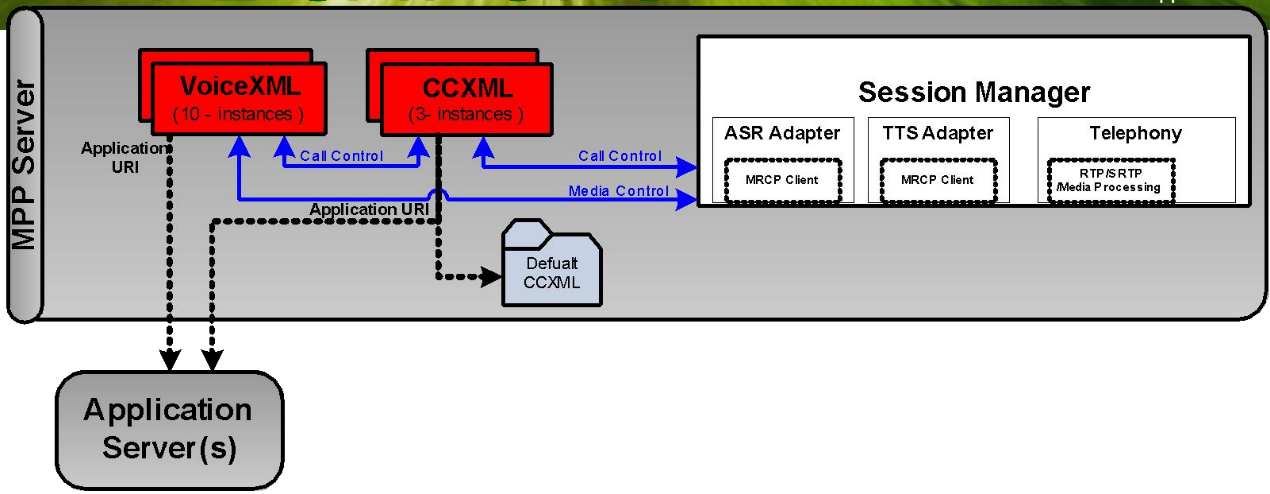

MPP supports running two types of custom application types:

- VoiceXML

  "…is a markup language for creating voice-user interfaces. It uses speech recognition and/or touchtone (DTMF keypad) for input, and pre-recorded audio and text-to-speech synthesis (TTS) for output."

- CCXML

  "…is an XML-based language designed to provide telephony call control support for VoiceXML and other dialogue systems. CCXML can control the setup, monitoring and tear-down of phone calls."

MPP – APPLICATIONS

- Even if only a VoiceXML application is configured, VP ***always*** runs CCXML (MPP uses its default.ccxml page).
- VPMS's *Application* page offers application URI combos…
  - VoiceXML (only option in VP 3.0)
    - MPP's *default.ccxml* page is used for call control
  - CCXML
    - CCXML does not require starting a VoiceXML page
      e.g. A CCXML app can just playing a prompt & disconnect
    - CCXML page can load a VoiceXML page
  - VoiceXML & CCXML
    - CCXML page can load a VoiceXML page
    - CCXML page can use session parameter to get the VoiceXML URI configured on the VPMS

```
<assign name=“vxmlappuri" expr="evt.appuri"/>
```

MPP – APPLICATIONS – CCXML? WHY DO I CARE?

- Replace CTI functionality
  - Advanced Call Control – Joining a single call to multiple VXML dialogs
  - Future – Handling conference calling and `<merge>`
  - Example – Find me/Follow Me
- Asynchronous eventing
  - Interrupt a running CCXML page.
  - Example: Caller waiting in queue for a Call Center Agent
- Outcall Web Services support
  - Advanced handling of call failures. Also, CCXML application can log failures to DB.
  - Future: CCXML could react differently if outbound call connected to fax machine or answering machine vs. person.
  - One LaunchCCXML could spawn multiple calls
  - Example: Call Blast
- Future – Data passed in-band data during call (not just at call creation)

MPP – APPLICATIONS – CCXML & VOICEXML FEATURES

Feature	VoiceXML	CCXML
Browser & Spec Version	• Avaya Voice Browser • Voice Extensible Markup Language (VoiceXML) Version 2.1. See http://www.w3.org/TR/2005/CR-voicexml21-20050613/	• Phonologies' Oktopous 1.1 • CCXML Version 1.0 (Working Draft), W3C, June 29, 2005. See http://www.w3.org/TR/ccxml/
Prompts	• Wav file Transcoding for supported types • TTS	• Wav file Audio encoding of the source prompt much match the audio selection for **MPP Native Format** on the VPMS)
Speech/DTMF	Supported	Not supported
Call Control	Has high level call control tags <transfer>	Low level call primitives such as accept, disconnect…
Configuration	Configure on VPMS	Not exposed
Caching	• Enabled • Must restart MPP to clear cache	• Disabled
Event Handlers	• Configurable via VPMS • Handler is merged into the running VXML app	• Configurable via VPMS • <transition> element in CCXML language
Application Logging	• `<log>` tags saved to ADR record on MPP and viewed in VPMS's ***Application Reports***. • VP customer privacy property supported `<property name="private" value="true"/>`	• `<log>` tags saved to ADR record on MPP and viewed in VPMS's ***Application Reports***. Note: ADR not written if VPMS is configured as VXML-only app. • VP customer privacy property supported `<property name="private" value="true"/>`
External Events from Web Svc	• Not supported	• Supported
Scripting	• ECMA script	• ECMA script "lite"

MPP – APPLICATIONS – MISC

- DD 6.0 supports creating CCXML &/or VoiceXML applications
- If you are creating CCXML & VoiceXML applications, be careful. The syntax can be different between the two.
- CCXML not 100% compliant
  - Supported:

    <accept>, <redirect>, <reject>, <createcall>, <join>, <unjoin>, & <disconnect>
  - Not Supported:

    <createconference>, <destroyconference>, & <merge>

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MPP START UP

1. **Linux Boots**
   - Linux inittab starts mppmon
   - Linux Init starts httpd daemon
   - Linux Init starts mpp daemon (mppsysmgr)
2. **mppsysmgr starts EventMgr process & waits for incoming requests from VPMS**
   - mppsysmgr sets state & config to *Stopped* & *No configuration*
3. **VPMS sends configuration**
   - Apache invokes MmsServer which forwards the configuration to mppsysmgr
   - mppsysmgr downloads Event Handlers & Certificates from VPMS
   - mppsysmgr sets state & config to *Stopped* & *Telephony configuration needed*
4. **VPMS sends start (administrator presses the Start button)**
   - Apache invokes MmsServer which forwards the start to mppsysmgr
   - mppsysmgr starts vxmlmgr, ccxml, and SessionManager
   - mppsysmgr changes state & config to *Running* & *Telephony configuration needed*
5. **VPMS sends Port Configuration**
   - Apache invokes MmsServer which forwards the port list to mppsysmgr
   - mppsysmgr sends port information to SessionManager
     - H.323 Ports: SessionManager registers the extensions with Gatekeeper
     - SIP Channels: SessionManager listens for incoming calls or waits for outbound call request
   - mppsysmgr changes state & config to *Running* & *Configuration OK*
6. **MPP ready to receive calls**

*Note: For simplicity, steps showing the VPMS sending a heartbeat request to the MPP have been skipped.

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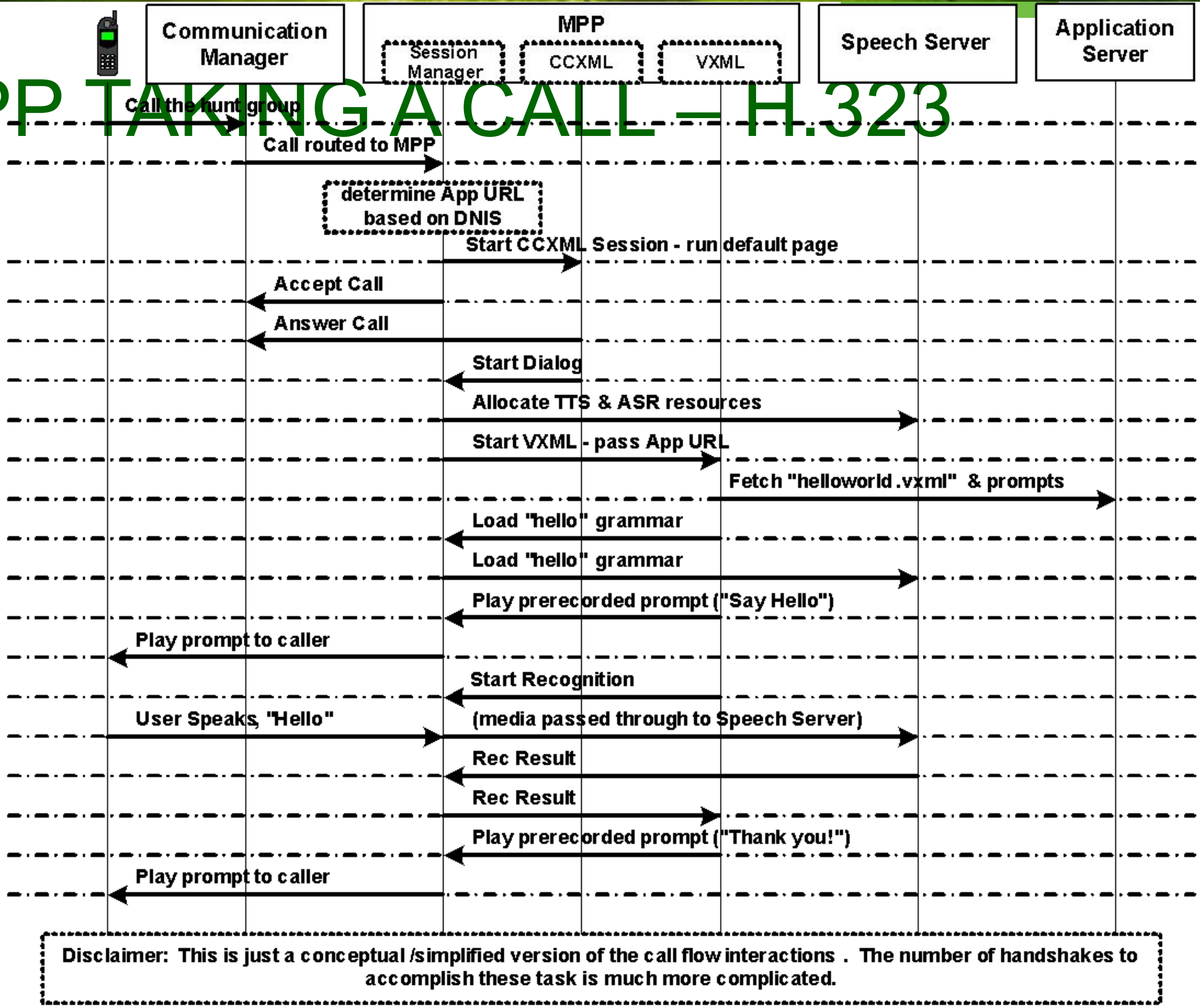
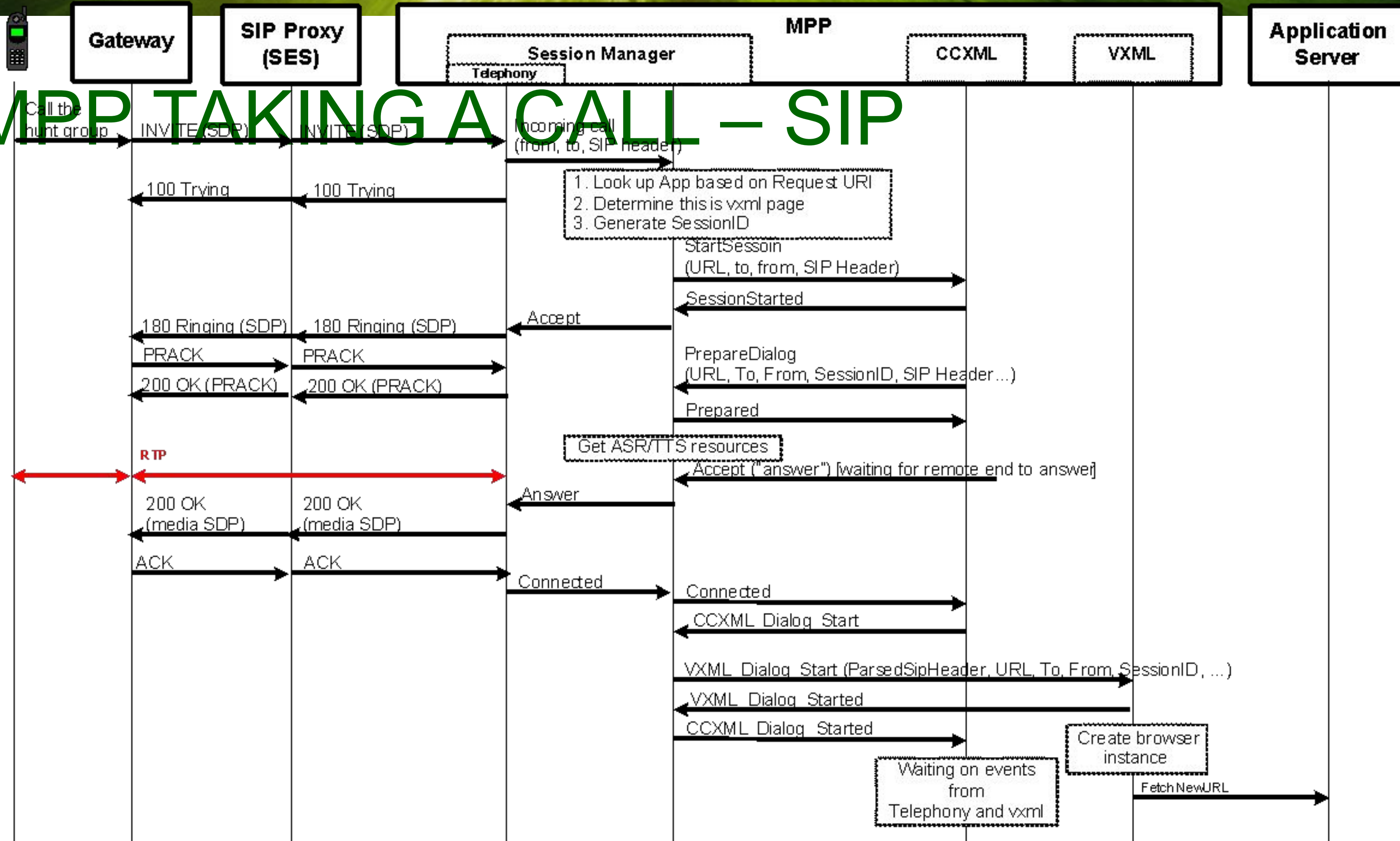
Simple VoiceXML “Hello World” application

```
<?xml version="1.0"?>
<vxml version="2.0" xmlns="http://www.w3.org/2001/vxml" xml:lang="en-us" >
<!–
          VoiceXML Say Hello Sample
-->
<!-- <menu id="Menu" dtmf="true"> -->
<menu id="Menu">
        <prompt  bargein="true">
         <audio src=“Sayhello.wav"> Say Hello </audio>
        </prompt>
        <choice dtmf="1" next="# ThankYou ">
        hello
        </choice>
</menu>
<form id="ThankYou">
        <block>
                <prompt bargein="false">
                        <audio src="ThankYou.wav"> Thank you </audio>
                </prompt>
        </block>
</form>
</vxml>
```

MPP TAKING A CALL – H.323



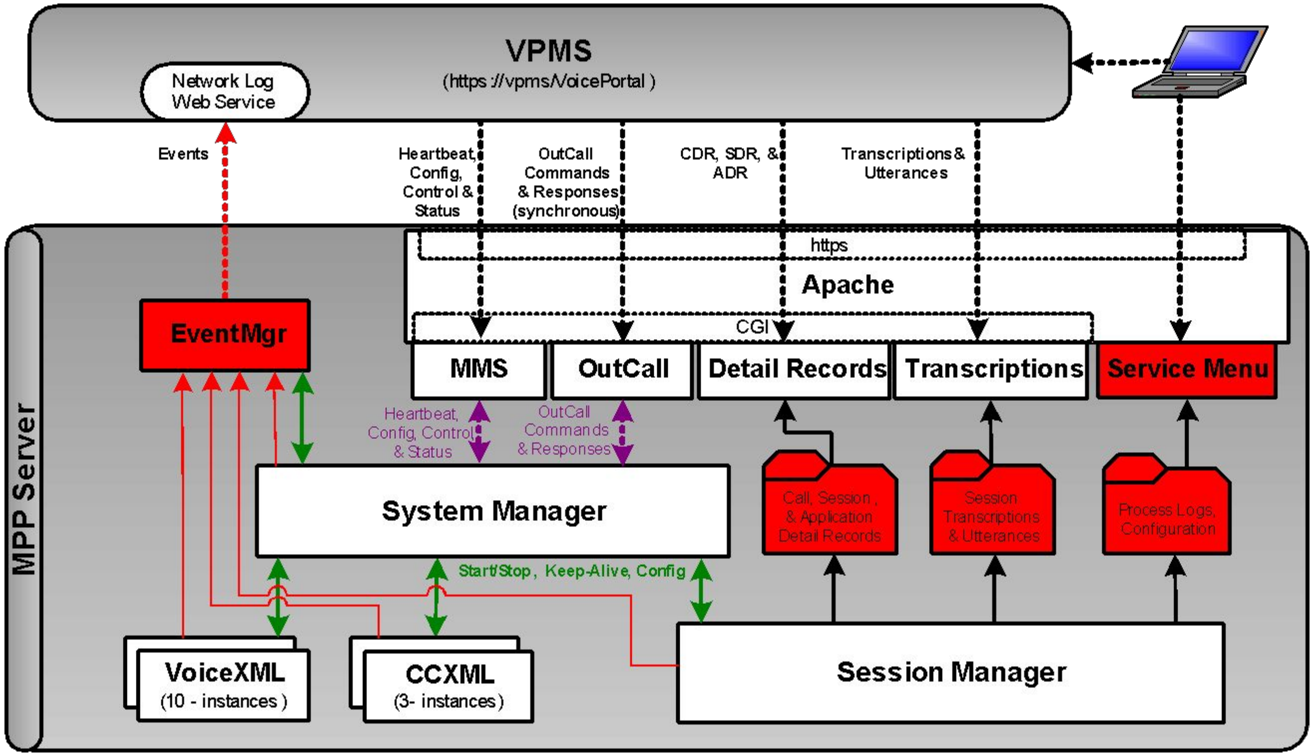
MPP TAKING A CALL – SIP



Disclaimer:

- SIP has many different implementation that can change the basic call flow between the Gateway, Proxy, and the MPP's Telephony provider.
- ASR, TTS (MRCP) is not shown. There are more interactions than those shown Some of the actual internal VP function calls were renamed for readability sake.

MPP – STATUS & DEBUG



- Records & Transcriptions
- Eventing & Tracing
- MPP Utilities

MPP –RECORDS

- Voice Portal provides detail Call Records so that you may…
  - Generate call load reports
  - Debug call quality problems
  - Debug application problems
  - & more!
- The MPP records 4 different Call Records types
  - Session Detail Records (SDR)
  - Call Detail Records (CDR)
  - Application Detail Records (ADR)
  - Transcriptions & Utterances
- VPMS provides various Reports to view Records
- Record files are purged from MPP based on VPMS configuration

MPP – STATUS & DEBUG– EVENTS & ALARMS

- **Events**
  - **ID format: P<*Subsys*>_<*Number*>**
  - **All severity levels written to the local MPP process logs.**
  - **Events are sent to the VPMS - configurable by severity level:**
    Fatal*, *Error* (default), *Warning*, & *Info
  - **Viewable in VPMS *Log Viewer***
  - **Some Events are aggregated before sent to VPMS**
  - **Debug Tip: Some MPP Events seen in *Log Viewer* report the Session ID. Use this Session ID when searching the MPP process logs.**
- **Alarms**
  - **Some Events trigger VPMS alarms.**
  - **ID format: Same as Event ID except starts with Q instead of P.**
  - **Viewable in VPMS *Alarm Manager***

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MPP – STATUS & DEBUG - TRACING

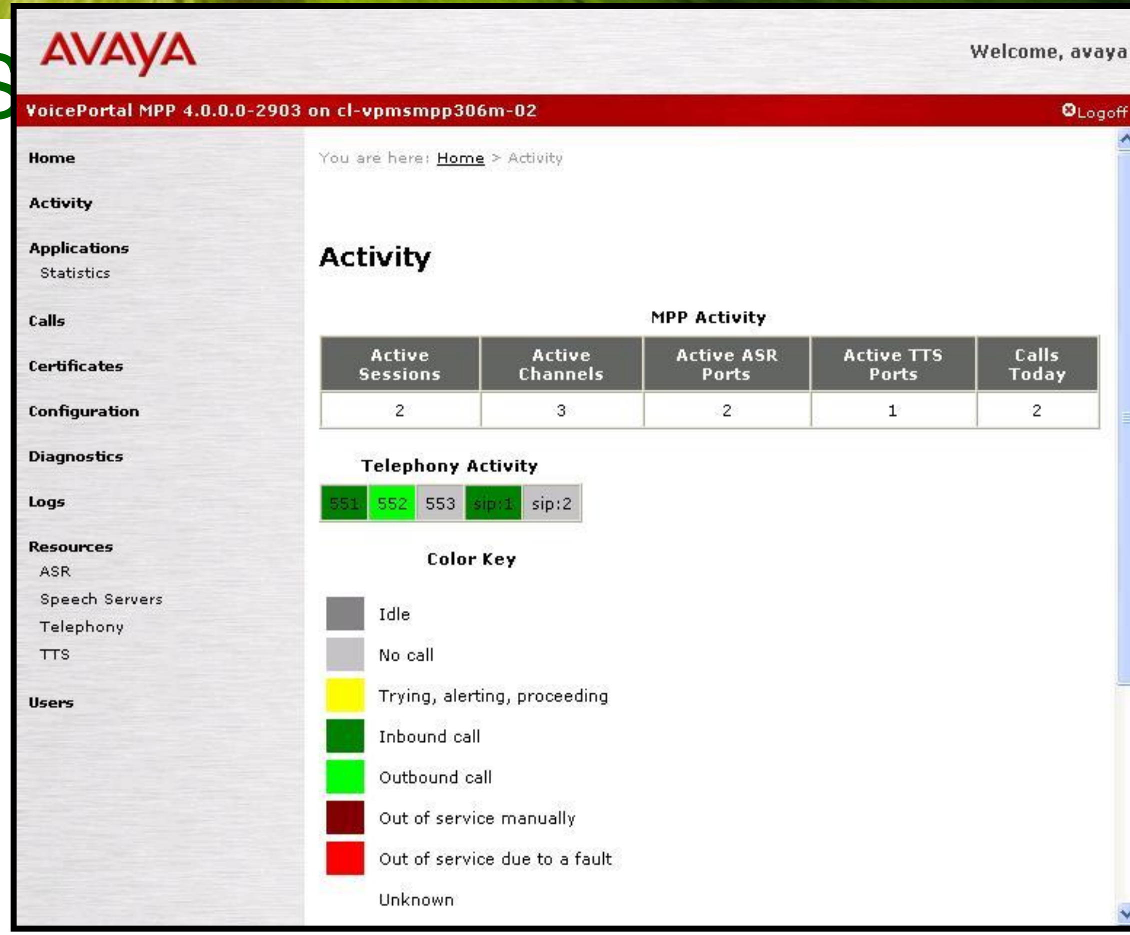
- **Intended for use for debugging problems.**
- **VPMS Configuration:**
  - **By level: *Off* (default), *Fine*, *Finer*, & *Finest***
  - **By Subsystem: *Telephony, System Manager, TTS…***
- **Only written to the local MPP process logs**
- **Debug Tips:**
  - **The logs aren't too helpful unless tracing is enabled**
  - **The logs with lots of tracing are difficult to dissect**
  - **Enabling tracing to *Finest* can cause performance degradation and is not generally recommended on production servers**
  - **Don't set *Telephony* to *Finest* unless Avaya Support tells you!**

MPP – STATUS & DEBUG – UTILITIES

- MPP Service Menu
  - web access for checking the MPP
  - Written in html/php
  - can be accessed by selecting the MPP from the VPMS’s *System Monitor* page and then clicking on the *Service Menu* link
- SSH/terminal window
  - Use the MPP’s scripts such as and to view status for processes and stations. See the Voice Portal’s Troubleshooting guide for a complete description of the available scripts.
  - Use basic linux commands such as *ps -e* and *top* for a list of system processes and system performance

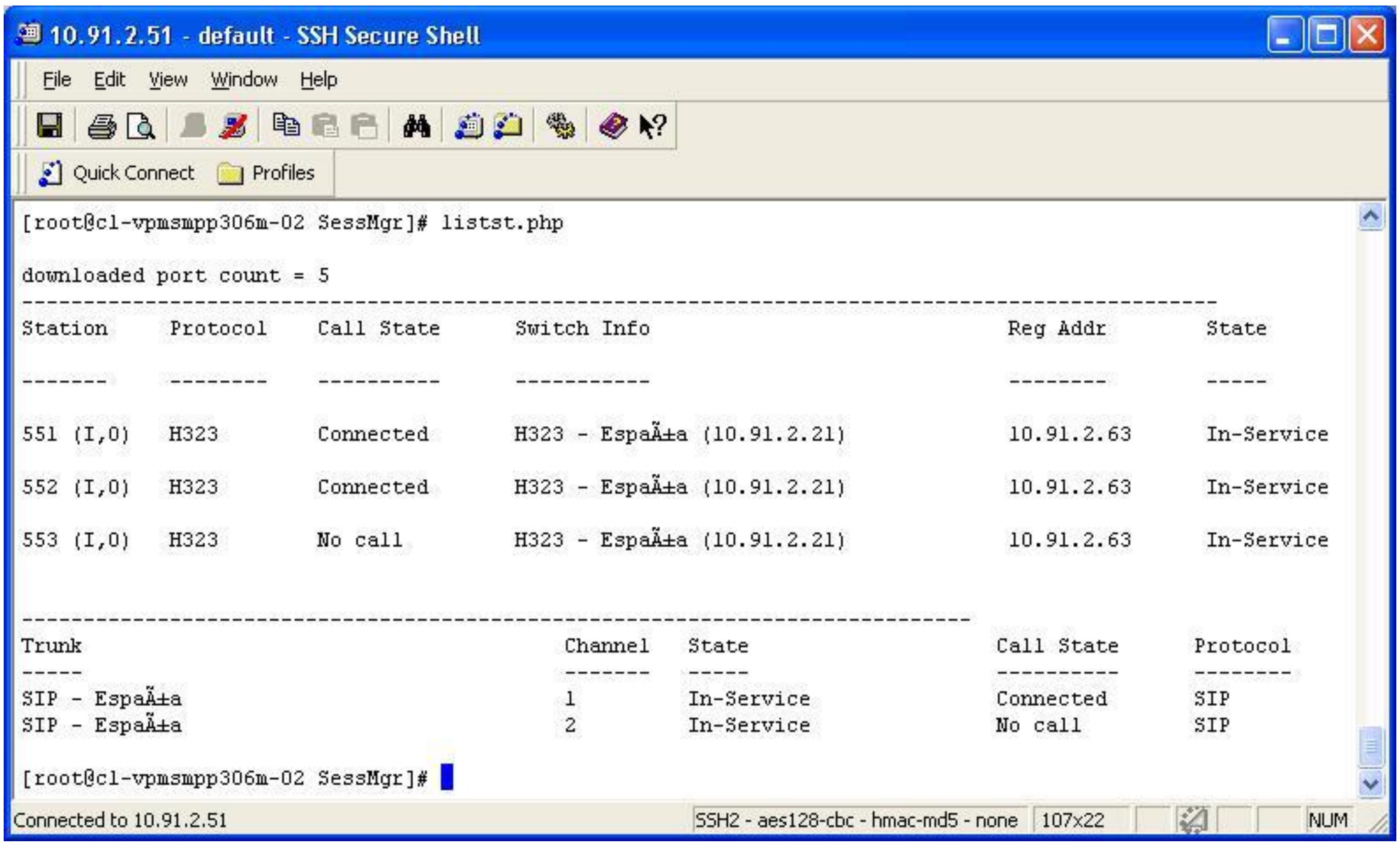
MPP – STATUS MENU

- Uses the same login credentials as VPMS.
- MPP service menu is not accessible if using tools such as: connect2, etc.
- Used to:
  - Resource status
  - View Configuration
  - View & Clear process logs
  - Run diagnostics & collect logs



MPP – STATUS & DEBUG – SCRIPTS

- If you cannot run the Service Menu, or just prefer the command line, almost all of the Service Menu functionality is available through the MPP's scripts.



FAILOVER AND DISASTER RECOVERY

- Failover
  - What happens when some component of the Voice Portal system fails.
  - Generally handled automatically by the system.
- Disaster Recovery
  - What happens when the entire Voice Portal system fails.
  - Generally handled manually.

FAILOVER IN VOICE PORTAL

- If the VPMS fails…
  - All calls in progress on various MPPs continue.
  - All MPPs continue to take new calls.
  - Log/alarm data generated by MPPs is lost.
  - No access to VPMS web user interface.
  - No SNMP notifications generated.
  - No response to SNMP queries.
  - No outcalls made by Application Interface web service.

FAILOVER IN VOICE PORTAL (CONT.)

- If an MPP fails…
  - All calls in progress on that MPP are lost.
  - Other MPPs are unaffected.
  - VPMS will redistribute telephony resources from dead MPP among the surviving MPPs.

    *Important – When you configure your MPPs in the VPMS, you must set the value of the field **Maximum Simultaneous Calls** appropriately in order for telephony resource redistribution to work properly.*

FAILOVER IN VOICE PORTAL (CONT.)

- If a speech server fails…
  - Calls in progress that are using failed speech server for ASR will fail.
  - Calls in progress that are using failed speech server for TTS will automatically switch to a different speech server.
  - All MPPs will avoid using failed speech server for new calls.
- If an application server fails…
  - All calls in progress that use applications on that application server will fail.
  - All new calls that use applications on that application server will fail.

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FAILOVER IN VOICE PORTAL (CONT.)

- If a gateway fails…
  - All calls in progress on that gateway will fail.
  - If an alternate gateway is configured, effected MPPs will re-register H.323 stations using alternate gateway and new calls will be processed normally.
  - If no alternate gateway is configured, no new inbound or outbound calls on the effected H.323 stations.
- If the SES fails…
  - All calls in progress on the SES will fail.
  - No new inbound or outbound calls can go through the SES.

DISASTER RECOVERY

- If an entire Voice Portal system fails, there is no way to automatically have new calls handled by a backup Voice Portal system.
- If the failed Voice Portal system gets licenses from a central WebLM server, licenses can manually be moved to a different Voice Portal system without obtaining new license file from Avaya.
- Depending upon your needs, you might partially/fully pre-configure backup Voice Portal system so that manual migration of calls after disaster strikes is easier.