



InterpreXer™, Phonologies VoiceXML IVR Server

Let your Communications Platform do the talking

Overview and Architecture

InterpreXer™ 4.1 is a SIP based VoiceXML (2.0 and 2.1) IVR system for Windows Desktop and Server environments that enables application developers to seamlessly develop and deploy VoiceXML applications. InterpreXer™ can recognize user input via DTMF and playback recorded audio in ulaw/wav formats. It also supports Speech recognition and Text-to-speech through third-party software using MRCPv2 protocol.

InterpreXer™ has been architected to separate media handling, document interpretation and telephony running as independent services to ensure high availability, scalability and module distribution. InterpreXer™ system consists of three main modules:

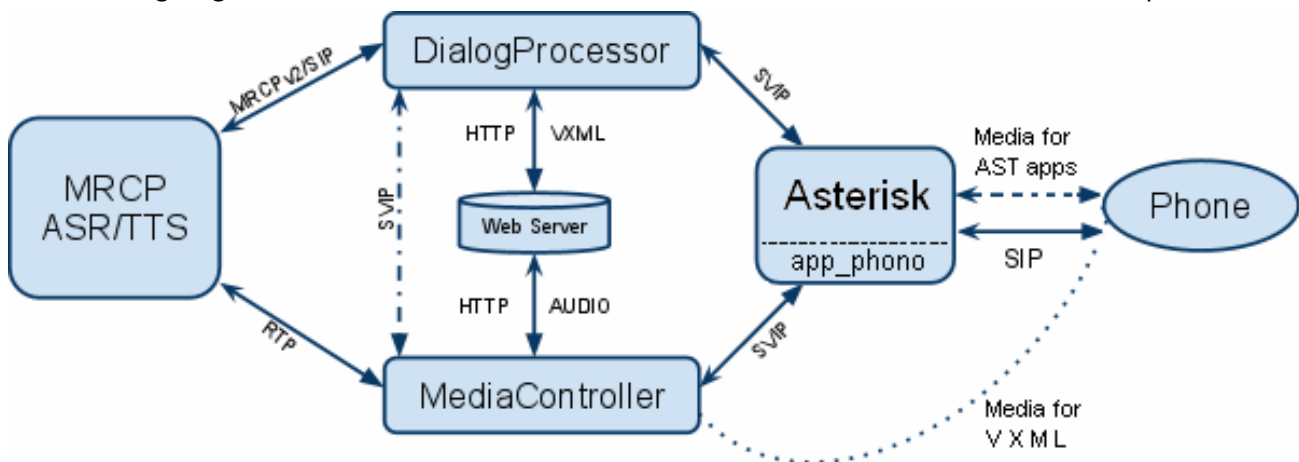
- DialogProcessor
- MediaController
- FreeSwitch® OR Asterisk®

DialogProcessor component is a VoiceXML Interpreter that downloads and interprets VoiceXML documents. It uses MediaController and FreeSwitch® OR Asterisk® for media and telephony related functions. It also does MRCP signalling for ASR and TTS functions.

MediaController is the media engine. It downloads and plays back audio, records user's audio to file system, routes the audio to ASR for recognition and streams MRCP TTS audio to the caller.

FreeSwitch® is a modular open source soft switch and Asterisk® is an open source PBX. The DialogProcessor and MediaController components interact with FreeSwitch® OR Asterisk®, as the case may be, through a custom-written module which is invoked via the dial plan of FreeSwitch® OR Asterisk® when they receive a call.

The following diagram describes the architecture and interactions between the three main components.



* The same architecture applies in the case of deployment with FreeSwitch®



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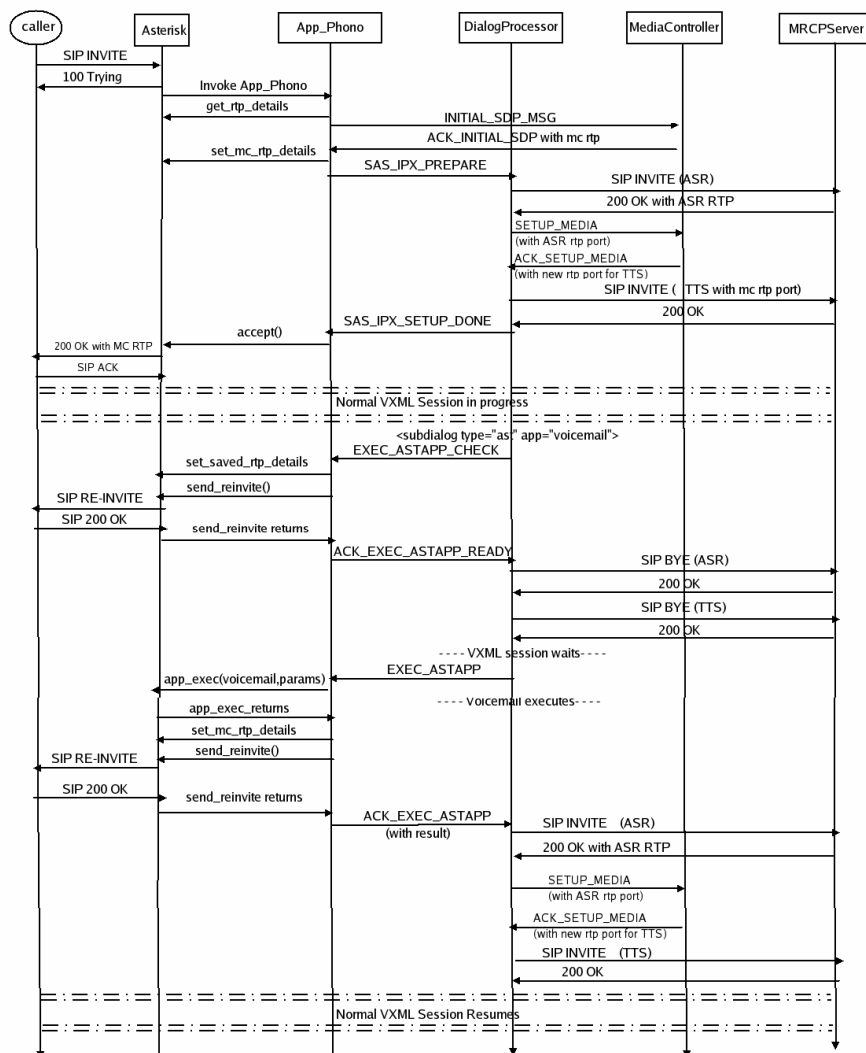
Custom Extension (* for Asterisk® only):

The InterpreXer™ can invoke Asterisk® Dialplan applications from within a VoiceXML session providing developers to use their existing Asterisk® applications at the same time. This is done by using the <subdialog> tag within your VoiceXML document as under:

```
<sub dialog type="asterisk" application="voicemail" data="extn_number">
```

Where type="asterisk" uses the Asterisk module app_phono to start execution of an Asterisk app, where application="voicemail" and sending custom information / data, like data="extn_number". InterpreXer uses a SIP re-Invite to start this session, releases the MRCP (SIP/Media) session with the ASR/TTS. Upon completion of execution of dialplan app the control is handed back to InterpreXer™ and the VoiceXML document execution resumes.

The following diagram describes the interactions between the components.





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Technical Specifications

Application Protocols:

- VoiceXML 2.0 & 2.1
- GrXML
- SSML

Speech Technology:

- ASR MRCP v2 (Loquendo, Nuance, LumenVox)
- TTS MRCP v2 (Loquendo, Nuance, LumenVox)

Signaling:

- SIP / RTP
- MRCP v2

Standards

- HTTP/1.1 & HTTPS
- ECMAScript

Operating System:

- Linux CentOS 5.x and RHEL 5.x
- Windows XP, 2003, 2008, Windows 7

Management GUI:

- Provisioning DID / SIP Extn
- Multiple vXML URI configuration
- Configuration of Speech Engines
- Realtime Logs

Phonologies InterpreXer™ 4.1 readily works with Media Gateways from CISCO / Linksys, Avaya / Nortel, ShoreTel, Allworx, Paetec, Quintium and Sangoma to name a few, and also SIP providers like Simple Signal, Star 2 Star, PortaONE and soon GoogleTALK & Skype.

About Phonologies

Phonologies is a provider of cloud based platforms to 'voice enable' business processes by cost effectively deploying web telephony, collaboration, unified communications, speech IVR and other types of applications within the social media, contact center and other enterprise domains. Our 'software-only' technologies are easy to plug-in to business processes and implement open standards that are flexible and easy to manage, resulting in interoperability, lower costs and protection of investments in the long run. Phonologies' flagship product, Oktopous™ powers over 30 Million calls worldwide each day. For more information on Phonologies, visit <http://www.phonologies.com>

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InterpreXer™ is also available for OEM implementations without the FreeSwitch® and Asterisk® modules.



Phonologies

The Voice of Technology