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.

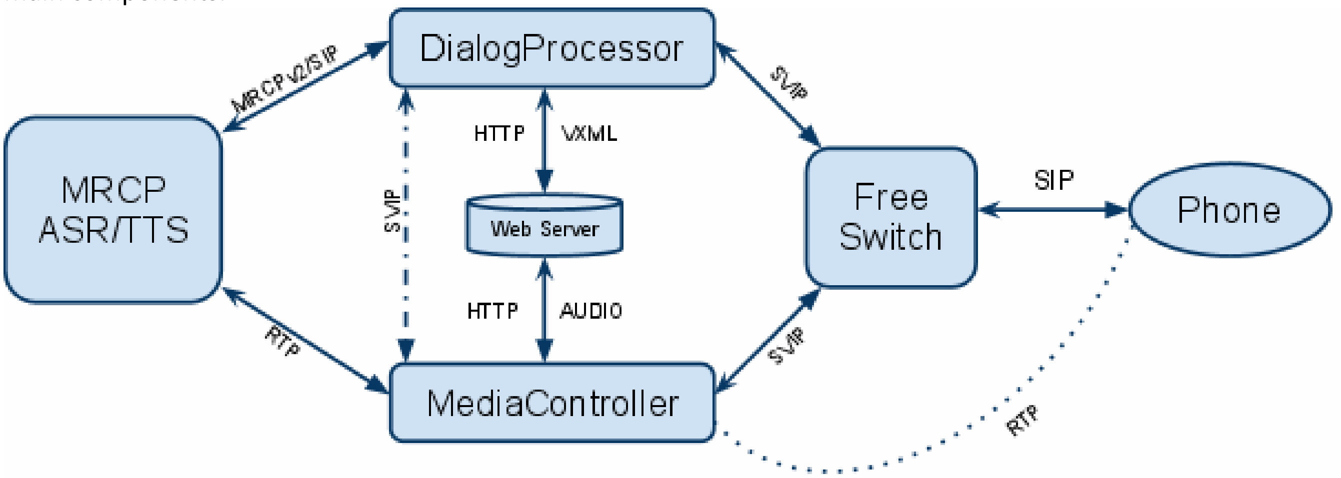
InterpreXer™ has been architected to separate media handling, document interpretation and telephony running as independent services to ensure high availability, scalability and module distribution. InterpreXer™ system consists of three main modules:

- DialogProcessor
- MediaController
- FreeSwitch

DialogProcessor component is a VoiceXML Interpreter. It downloads and interprets the VoiceXML documents. It uses MediaController and FreeSwitch 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. The DialogProcessor and Media Controller components interact with FreeSwitch though a custom-written module, which is invoked when FreeSwitch receives a call based on the dial plan. The following diagram describes the architecture and interactions between the three main components.



InterpreXer™ is also available for OEM implementations and embedding on Telephony Boards that cater to 8, 16, 24 port size deployments.

InterpreXer™, Phonologies VoiceXML IVR Server

Let your Communications Platform do the talking

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 Skype, Simple Signal, Star 2 Star, PortaONE and soon GoogleTALK.

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 5 Million calls worldwide each day. For more information on Phonologies, visit http://www.phonologies.com

Contact Information:

Email: InterpreXer@Phonologies.com
Phonologies (India) Private Limited
Ph: +91.22.27684560 or +91.40.27018993

InterpreXer™ is also available for OEM implementations and embedding on Telephony Boards that cater to 8, 16, 24 port size deployments.