VXI* VoiceXML browser

Open standards IVVR platform for the Next Generation Voice & Video Applications

Software-based and running on Linux open networks interfaces using IP / 3G media processing full controlled by VoiceXML

VXI* VoiceXML browser gives operators and service providers the ability to rapidly develop and deploy new, innovative VoiceXML-controlled voice and video applications in IP, PSTN, and 3G-324M networks. VXI* is fully compliant with W3C VoiceXML 2.0 and some 2.1 specifications, and can easily be integrated into automatic speech recognition[1] (ASR) and text-to-speech[2] (TTS) systems to enable advanced IVVR applications, voice and video communications, and real-time video calling applications. The VXI* VoiceXML browser can be installed in common hardware configurations, providing a highly-scalable and highly available software base system to meet all customers' business and technical VoiceXML requirements.

1 – release 3.0 or later is required
2 – all releases 2.X and 3.0

VXI* VoiceXML interpreter is pluggable into a standard Digium Asterisk PBX release. This feature gives important benefits of scalability and low cost solution profile. Most users of this popular Open Source PBX can run now VoiceXML in the same server.

Key features

- Carrier class implementation
- VoiceXML 2.0+ compliant (with some 2.1 tags added)
- HTTP full supported and HTTPS
- HTTP outbound dialing API available
- E1, T1, IP connections
- Asterisk *CLI> full integration
- Web enabled CDR console

Voice (IVR)

- Up to **150 audio VoiceXML sessions** per CPU
- SIP/SS7/ISDN network
- Speech integration with Asterisk PBX

Video (IVVR)

- Up to **60 video VoiceXML sessions** per CPU
- 3G-324m network
- Video play and record / .3gp format supported

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Product description

VXI* VoiceXML browser is a voice and video interactive response system (IVVR) such as IVR. All developers can easily deploy end user dialogs. Our solution is W3C VoiceXML 2.0+ compliant, full 2.0 tags and some 2.1 tags are supported like useful <transfer> and can be used to build VoiceXML-controlled services in SIP, ISDN (TDM) and 3G (3G-324m) phone networks. With VXI* you have several addons extensions like the Outbound clic2call API and most of Asterisk's complementary components are supported too.

Turn-key solution

Whole solution is available as a 100% packaged software, easy to install in a common Linux 32bit OS server as Debian, Suse, Ubuntu, Redhat distributions... Main servers configurations can be used to install your VoiceXML solution base on VXI*. You can buy yourself hardware to any vendor that support Linux environments and setup a VoiceXML IVVR system for your needs.

Hi-scalability

The remote *CLI> management console allows administrators to monitor the VoiceXML browser, start and stop services, and stop taking new calls so that the server can be brought offline for maintenance without affecting connected callers. VXI*can manage unlimited hosting accounts with different channels capacity to make easier your VSP customer management and VoiceXML resources sharing.

Voice IP/SS7 applications

VXI* allows to get speech-enabled IVR standard voice applications like voice messaging, auto-attendant services, conferencing. Using TTS voices is supported with an advanced universal HTTP open source interface that help our customers to manage multi-vendor speech engines in same or different servers. VXI*supports VoIP network with all Asterisk PBX open source code supported by the most important worldwide telephony developer community. Digium TDM boards are cheap and simple to install in your system to add E1/T1/SS7 connections.

Video IP/3G applications

The video configuration supports video calls by playing and recording .3GP format (audio and video channels). This feature allows to build any interoperability video portal for IP/3G services . VXI* connect to 3G-324m Networks using a Digium TDM E1/T1 cards without using any a specific gateway.

Hardware examples

VXI* can be installed in common 1U server like (only examples):

- HP Proliant DL140
- HP Proliant DL320
- Fujitsu Siemens Primergy
- DELL Poweredge

Any other vendors/series with similar hardware can be used to build your VoiceXML server system.

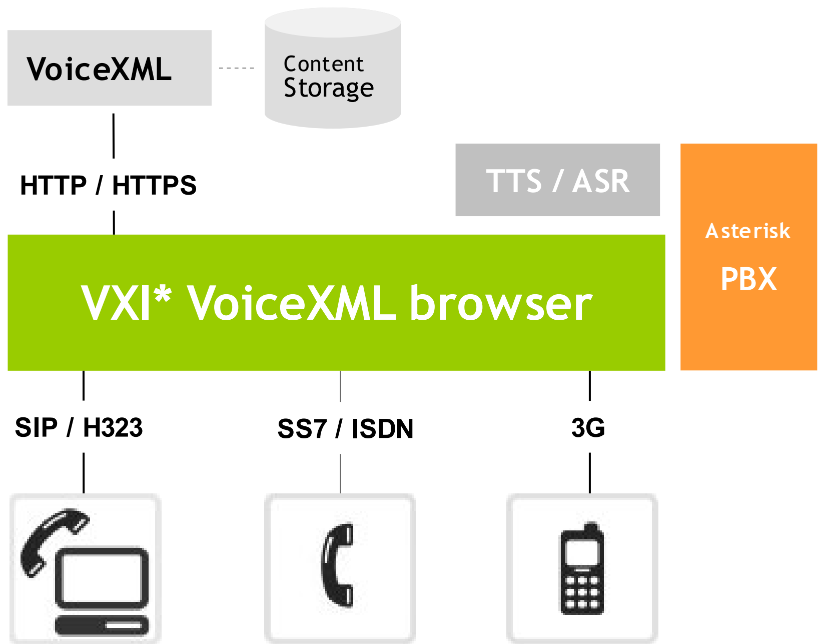

Figure 1 – VXI* interfaces

Technical specificacions

VoiceXML support

VoiceXML features

- VoiceXML 2.0 compliant
- extended VoiceXML 2.1 tags
- VoIP / SIP control interface
- VoiceXML fetched HTTP or HTTPS
- VoiceXML scripts are cached
- Detailed logging with full information level of the voice browser

VoiceXML compliance

- **VoiceXML**: W3C Voice Extensible Markup Language Version 2.0 and 2.1
- **SSML**: W3C Speech Synthesis Markup Language Version 1.0
- **SRGS**: W3C Speech Recognition Grammar Specification Version 1.0
- **TTS**: VXI universal connector with an HTTP hypercache interface
- **ASR**:VXI Recognition interface using the Lumenvox's Speech Engine for Asterisk interface
- **OpenSER:** SIP application server integration full supported
- **Asterisk:** PBX SIP/SS7/TDM Digium board hardware/drivers supported

System management

- Linux operating system kernels Debian, Suse, Ubuntu, Redhat like distributions with GCC+ 4 libs supported
- 32 bit binary packages, for 1,4, 2,X, 3,X last releases
- Asterisk terminal console (*CLI>) for command line monitoring
- FTP / SSH protocol supported

Special extensions

- Outbound HTTP API, HTTP/PHP extension that provide cli2call.
- Call Detailed Reports (CDR) extension, HTTP web enabled for Asterisk PBX
- Outbound predictive dialer (not included)
- VCR audio function over VoiceXML for WAV clips with property "control" (* and # keys) included since 2.2. release
- Google Talk , jabber interface supported

Media formats

Audio file formats

- RAW 8kHz 16-bit µ-law mono
- GSM 8kHz 16-bit mono
- WAV 8kHz 16-bit (PCM) mono
- MP3 Mpeg audio layer 3
- Asterisk audio files supported

Video file formats

- 3GPP .3GP Basic Profile
- JPG image file format

Media support

Speech Recognition

- Uses VXI/Asterisk protocol to interface the VoiceXML Server to speech recognition (ASR) servers

Speech Recognition Engines

- Lumenvox (Speech Engine) – since release 3.0 with ASR connector included.

Speech Synthesis

- Prompt Engine provides low-latency streaming of synthesized speech to bearer channels
- Synthesized prompts are cached for improved performance and reduce TTS usage
- Uses SSML tags to improve speech articulation

Speech Synthesis Engines

- Flite (freeware) included with VXI*
- Cepstral
- Loquendo
- Nuance - Scansoft
- Neospeech
- Verbio
- Acapela

Voice and video support

Voice channel formats

- GSM 8kHz 16-bit mono
- WAV 8kHz 16-bit (PCM) mono

Video specifications

- Supports 3GPP 3G-324M standards
- Video play and record, including prompts, announcements, and messaging
- Video codec support: 3GPP, H.263, H. 263+, H.264
- 3G-324m terminal support
- Video streaming using RTSP1
- TTV Text-to-Video
- Video silence generation
- Background Recording all channels
- Video conferencing up to 12 channels

Video channel formats

- H.263 Baseline Level 10
- H.263+, H.264

Voice specifications

- Supports SIP ((Session Initiation Protocol), H323 VoIP
- Voice recording in GSM/WAV (PCM)
- ISDN E1/T1
- T30 Fax
- Voice codecs G.711, A-law, µ-law, G.729
- Audio conferencing
- TTS Text-to-Speech Multi language
- VSD Voice-Silent Detection
- Voice silence generation

Voice channel formats

- SIP, H323
- IDSN/SS7 with Digium TDM boards

Voice codecs

- G.711, A-law, µ-law
- G.726 @ 32 kbps (DSP)
- G.729.ab
- AMR

Video codecs

- AMR narrow band

I6NET Solutions and Technologies Limited is an European company dedicated to research and development of telecommunications and Internet technology. We are advanced services experts in voice and video interactivity in line with the latest evolution in telephony.

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