Vxi* VoiceXML browser

Datasheet

Use the power of VoiceXML, Asterisk® and open network interfaces to run IP media solutions on Linux

The VXI* VoiceXML browser gives operators and solution providers the ability to rapidly develop and deploy innovative voice and video applications via VoIP, PSTN and 3G-324M networks. VXI* is fully compliant with the W3C's VoiceXML 2.1 specification and is integrated with automatic speech recognition (ASR) and text-to-speech (TTS) software to enable advanced voice and video solutions, and real-time video calling applications. VXI* can be installed in common hardware configurations, providing a highly scalable base system to meet all customers' business and technical VoiceXML requirements.

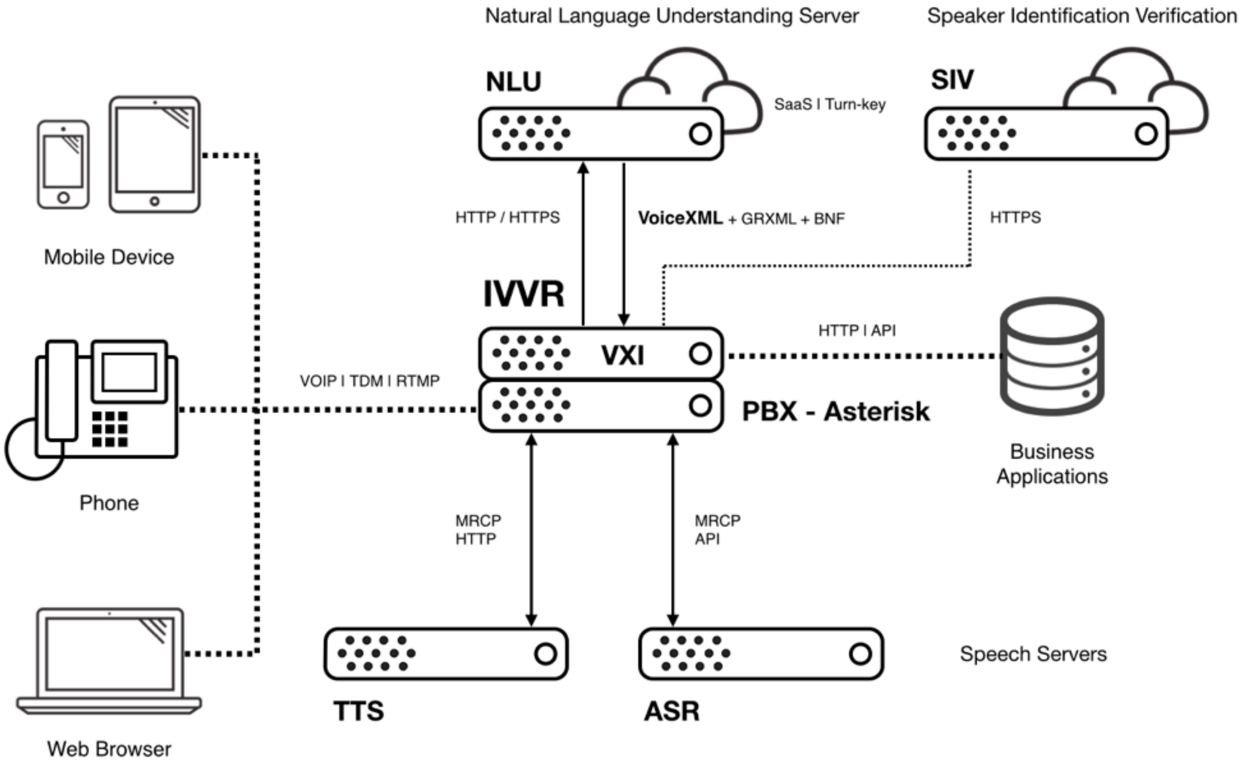
Features:

- Software-Based Solution, 100% packaged
- Available for many Linux OS distributions for 32bit or 64bit
- Simultaneous voice & video application interoperable
- 3G-324M support that includes H.263, H.263+, H.264
- VoiceXML 2.0 and 2.1 compliant with some extended tags added
- Real-time any-to-any video transcoding and rate adaptation
- Webcall over any web browser with Flash Player
- Mobile Videocall RTMP interface for iOS and Android devices
- Xtras* addons available to extend your VXI* platform

Software Partner

Architectural Model

The VXI* VoiceXML interpreter works directly with the Asterisk PBX software supported by Digium®. Not only can users of the open source PBX run VoiceXML applications in the same server, they can now offer these powerful, scalable IVR / IVVR solutions at an affordable cost.



IVR	Interactive Voice Response
IVVR	Interactive Voice and Video Response
ASR	Automatic Speech Recognition
TTS	Text-to-Speech
PBX	Private Branch Exchange
NLU	Natural Language Understanding
SIV	Speaker Identification Verification
PSTN	Public Switched Telephone Network
VOIP	Voice over IP
TDM	Time-Division Multiplexing
MRCP	Media Resource Control Protocol
HTTP	Hypertext Transfer Protocol
HTTPS	Hypertext Transfer Protocol Secure

Voice and Video Applications

The VXI* VoiceXML browser is commonly used for IVR or IVVR applications where voice and video combine for dynamic solutions with easily deployed end-user dialogs. VXI* is W3C VoiceXML 2.1 compliant, with some extended tags like <transfer> also supported. The browser is used to build VoiceXML-controlled services via SIP, H323, Flash/RTMP, ISDN (TDM) and 3G (3G-324m) phone networks. In addition, there are several Xtras* addons extensions like the Outbound Dialer and most of Asterisk's complementary components.

Ready for any IVR / IVVR developments

Mobile Services

- Mobile TV
- Video Sharing
- Video eLearning
- Video Surveillance

Carrier Services

- Voice and Video Messaging
- Auto-Attendant
- Networks Gateway
- Customer Care
- QoS Surveys

Entertainment Services

- Video Portal
- Televoting
- Video dating

Contact Center Services

- Voice and Video Self Services
- Predictive Dialing
- Emergency Notification
- Voice and Video Broadcast

Network Gateway configuration

- Gateway SIP | H323 < > 3G-324m
- Gateway SIP | H323 < > Flash/RTMP
- Video Call Recording over SIP | H323 | 3G-324m | Flash/RTMP
- Video Call Conferencing over SIP | H323 | 3G-324m | Flash/RTMP

Direct Inward Dialing (DID) phone numbers forwarding

- DID ISDN | SIP for Video 3G-324m
- DID ISDN | SIP for Video Flash/RTMP

Product Description

Turn-key Solution

VXI* is available as a 100% software packaged solution – easy to install in a common Linux 32bit / 64bit OS server with Debian, CentOs, Suse, Ubuntu, and RedHat distributions. VXI* solutions can be run on the main server configuration and on any hardware that supports Linux, enabling the developer to build a custom-fit system for their needs.

Hi-Scalability and System Management

VXI* / Asterisk 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 channel capacity to make it easier for VSP customer management and VoiceXML resources sharing.

Speech Interfaces

VXI* allows speech-enabled (ASR) voice applications like voice messaging, auto-attendant services, and conferencing. VXI* includes several ASR connectors with Asterisk native API or MRCP v1 and v2 through uniMRCP. Text-to-Speech (TTS) is also supported with an advanced, universal HTTP open source interface that helps developers manage multi-vendor TTS engines in the same or different server(s).

Voice Applications

VXI* works on Voice over IP networks (SIP/H323) with the Asterisk open source code supported by a well-known, worldwide telephony developer community. Developers can use Digium's TDM hardware as an affordable and simple way to add E1, T1 and SS7 connectivity.

Video Applications

The video IP/3G 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.

Xtras* addons to extend your platform

VXI* can be extended with a large number of Xtras* addons to provide specific features to VXI*. Some addons are: Video transcoders, Flash/RTMP channel server, Outbound Dialer, Outbound API, Call Recording, Video Conferencing… etc

Virtualization for Cloud Telephony

VXI* is only software base and can run over cloud or virtual servers.
Now, you can setup your platform over Amazon EC2 servers. I6NET provides Platform as a Service (PaaS), if you require to run your own services without IT investments.

Technical Specifications

System Features

- Simultaneous support of PSTN and IP calls
- Simultaneous support of audio and video
- Seamless migration of PSTN to IMS
- Flexible configurations supporting audio only, audio with video
- Outbound and/or inbound call control through Asterisk
- Audio playback support, with VCR controls
- Video transcoding through optional Video Transcoder server
- Audio conferencing support for contact-center-like functions
- Capacity upgrade through software license

Web Protocols

- VoiceXML 2.0 and 2.1
- MRCP v1 and v2
- SRGS 1.0
- SSML 1.0
- RTSP
- RTMP
- HTTP and HTTPS

VoiceXML Features

- VoiceXML 2.1 compliant and some additional tags supported
- VoiceXML scripts are cached
- Detailed logging with full information level of the voice browser
- SIP interface allows load balancing and redundant configurations

Speech Recognition

- VoiceXML server control of speech server through Asterisk API
- VoiceXML server control through uniMRCP v1 and v2
- Application-selected ASR

Text to Speech

- Synthesized prompts cached for improved performance (hypercache)
- Prompt engine provides low-latency streaming of synthesized speech-to-bearer channels
- Application selected TTS
- Synthesized prompts are cached for improved performance and reduce TTS usage
- Uses SSML tags to improve speech articulation for TTS engines SSML compliant
- HTTP technology allowing multi-server resource sharing between different VXI* systems.

Media Processing

- G.711 A-law/µ -law, AMR-NB, G.723.1, G.726 @ 32 kbps (DSP), G.729ab (Digium License)
- H.263, H.263+, H.264, MPEG-4 , H263 Sorenson (Flash/RTMP)
- On-demand, any-to-any, real-time video transcoding supported through VXI* video transcoder
- DTMF detection and generation
- Echo cancellation through Asterisk
- Voice Activity Detection
- Audio transcoding
- Audio ringback tones over VoiceXML
- VCR controls (audio and video)

Voice and Video Conferencing Support

- Conferencing support through Asterisk control
- Echo cancellation through Asterisk control
- Coaching mode
- Call recording
- Music-on-hold

Technical Specifications

Video Support

- Video support through VoiceXML
- Optional embedded 3G-324M gateway (Xtras* Video IP/3G addon)
- H.264, MPEG-4, H.263 with any-to-any video transcoding
- Video rate & size adaptation (43Kb to 384Kb, QCIF/CIF)
- Video refresh (RFC 5168)
- Integrated with ASR and TTS support
- Separate audio or video sources
- Video conferencing up to 12 channels
- Background Recording all channels
- Simultaneous video play and record (video karaoke)
- Streaming video with RTSP connection sharing (Xtras* Video IP/3G addon)

Application Management

- Provisioned through CLI* Shell Management Console
- DNIS to application URI mapping

Supported ASR Servers

- ScanSoft / Nuance
- Loquendo
- Verbio (Native API / MRCP)
- Lumenvox (Native API / MRCP)
- Vestec (Native API / MRCP)
- Voice Interaction (Native API)
- MRCP v1 and v2 compatible engines

Supported TTS Servers

- ScanSoft / Nuance (Realspeak, Rhetorical)
- Acapela
- Loquendo
- Ivona
- Verbio
- Voxygen (Baratinoo)
- Cepstral
- Voice Interaction (Audimus)
- Flite / eSpeak / MBROLA (free TTS)
- Yantra Software
- Google Text-to-Speech

Supported Streaming Servers

- Helix Mobile Server (Real)
- Darwin Streaming Server (Apple)

Media Formats

- .raw 8kHz 16-bit (PCM) mono
- .gsm 8kHz 16-bit mono
- .wav 8kHz 16-bit (PCM) mono
- .mp3 (MPEG audio layer 3)
- .3gp (324M video)
- .tiff (TIFF image file format)
- And any other Asterisk's audio file format supported

Media Interfaces

- FILE:// local or remote
- HTTP(S):// local or remote, with caching per header
- RTSP:// local or remote, streaming audio or video

Technical Specifications

Media Transport
- RFC 3550/3551 (RTP)
- RFC 2326 (RTSP)
- RFC 2833 (DTMF)
- RFC 3267/IF2 (AMR)
- RFC 2190 (H.263)
- RFC 2429 (H.263+)
- RFC 3016 (MPEG-4)
- RFC 3984 (H.264)
- RFC 5552 (SIP-VoiceXML)
- RTMP

Network Interfaces
- Simultaneous PSTN and VoIP support with integrated PSTN-SIP gateway
- Gigabit Ethernet (redundant recommended)
- T1/E1 Through Asterisk compliant boards
- ISDN Through Asterisk compliant boards
- ISUP/SS7 Through Asterisk compliant boards
- ISUP/SIGTRAN Through Asterisk compliant boards
- SIP (internetworking with ISDN/ISUP/BICC)
- RTMP Flash
- H323 for voice and video
- T30 FAX

Capacity per server, audio
- VoiceXML SIP | PSTN — 480 ports
- Capacity upgrade through software license

Capacity per server, video
- VoiceXML SIP | PSTN — 240 ports
- Capacity upgrade through software license

Video Transcoder Capacity (per unit)
- Capacity depends on codec, frame rate, bit rate, and size. For example:
  Up to 180 streaming ports of H.263 to H.264 (QCIF) transcoding

Hardware Servers (examples)
- HP DL360 G6
- HP DL160 G6
- HP DL320
- HP DL120
- DELL Poweredge series
- Fujitsu Primergy series
- Any other Linux Server compliant hardware, min 2Gb or 4Gb RAM

Asterisk compliant kernels
- Asterisk 1.4.X 32bit / 64bit
- Asterisk 1.6.X 32bit / 64bit
- Asterisk 1.8.X 32bit / 64bit
- Asterisk 10,11 32bit / 64bit (coming soon)

Linux Distributions
- Linux Debian 7.0, 6.0, 5.0, 4.0, 3.0 32bit / 64bit
- Linux CentOS 6.3, 5.4 32bit / 64bit
- Any other compliant distribution like Ubuntu, Readhat, Mandriva, Slackware, Fedora...

www.i6net.com

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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