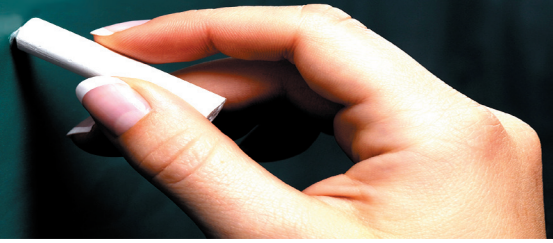


As Easy As...

123

Voxeo Prophecy
Voice Platform



Why does telephony have to be rocket science? We couldn't think of a reason, either. That's why Voxeo created Prophecy, an on-premise voice platform that's so easy you won't stop talking about it.

Easy to Get.

Prophecy is available via one-click download to anyone with an Internet connection. No NDA. No sales guy. No hassle. And the **first two ports are free** for use in development and live deployment at www.voxeo.com/prophecy, so you don't even need a budget to get started.

Easy to Support.

No platform is an island. Prophecy's robust support of standards means you'll have no worries about proprietary lock-in or difficult integration with the rest of your systems. Prophecy's **VoiceXML 2.0-certified browser** is the only platform that passes every single mandatory and optional VoiceXML forum certification test, so you never have to worry about missing features. Prophecy also includes a full **CCXML 1.0 browser** for call control and a 100% **SIP-compliant** telephony layer. To complement the built-in Prophecy **speech recognition** and **text-to-speech engines**, full support of **MRCP** enables easy integration with other leading speech engines.

Easy to Use.

You can download and install Prophecy in under two minutes. Prophecy comes ready-to-go with a host of pre-configured components, including a **soft phone, ASR and TTS engines, database engine,** and **web server** supporting **PHP 5.1** and **Java/JSP**.

Easy to Deploy.

With advanced features such as **outbound calling, n-way conferencing,** and **call recording** built right in, Prophecy makes it easy to deploy sophisticated applications on a simplified architecture. What's more, open source applications like an **auto attendant, voicemail,** and **conference management** solutions are included. Prophecy is also seamlessly compatible with Voxeo's hosted VoIP and IVR offerings, making it easy to gain access to **mission-critical redundancy** and **burst capacity** by fusing premise and hosted systems.



-This space intentionally left blank-

(After all, you can download the software faster than you could have read this section)



www.voxeo.com/prophecy

Prophecy Voice Platform Specifications and Features

Major Details

- 100% VoiceXML-compliant
- World's most proven CCXML
- 100% SIP-based telephony, works out-of-the-box with most VOIP providers
- Built-in high-quality speech recognition
- Built-in high-quality speech synthesis
- Built-in conferencing, up to 50 callers
- Built-in call recording
- Built-in PHP 5.1 application server
- Built-in Java/JSP web application server
- Built-in SQLite database engine
- Built-in SIP soft phone
- Runs on Windows 2000, XP, and 2003
- Linux and Mac OS X support coming soon

Call Control Features

- SIP inbound/outbound call support
- SIP call redirection (SIP 302 status)
- SIP call rejection
- SIP call routing and transfers
- SIP call leg bridging
- SIP REINVITE audio re-routing
- SIP registrar server support
- SIP proxy server support
- SIP authentication support
- Configurable SIP port range
- NAT IP translation support
- Multi-ethernet-card bridging support
- Runs on standard x86 platforms

Media and IVR Features

- Audio prompt/announcement playback
- Audio bridging

- Audio mixing/conferencing
- Audio noise removal
- Audio fixed gain control
- Audio dynamic gain control
- Audio voice activity detection (VAD)
- Audio call recording
- Automated speech recognition (ASR)
- Text-to-speech (TTS)
- Audio playout jitter buffer
- DTMF tone detection
- DTMF tone generation
- Configurable RTP port range
- NAT IP translation support
- Multi-ethernet-card bridging support
- Supports VoiceXML 2.x IVR
- Supports CallXML 3.0 IVR
- Runs on standard x86 platforms

Compatibility Features

- Works with Level(3) SIP
- Works with Broadvoice SIP
- Works with Delta3 SIP
- Works with SER/OpenSER SIP
- Works with Asterisk SIP
- Works with Cisco SIP
- Works with Lucent SIP
- Works with Sipura SIP
- Works with Sonus SIP
- Works with most other SIP services

IETF Standards Support

- RFC 3261 SIP
- RFC 3310 SIP Authentication
- RFC 1889 RTP Media
- RFC 1890 RTP Audio

- RFC 2327 SDP
- RFC 3264 SDP negotiation
- RFC 2833 DTMF and events
- RFC 3263 SRV DNS records
- RFC 3761 ENUM URI DNS records
- RFC 3764 ENUM SIP DNS records
- RFC 3164 UDP Syslog logging
- RFC 3195 TCP Syslog logging
- RFC 2865 RADIUS metering
- RFC 2616 HTTP protocol
- RFC 2617 HTTP authentication
- RFC 2964 HTTP state management
- RFC 2965 HTTP state management
- RFC 3927 Dynamic IP config
- RFC 2136 Dynamic DNS updates
- DNSEXT DNS Service Discovery Draft
- MMUSIC RTSP Draft
- SPEECHSC MRCP Draft

W3C Standards Support

- VoiceXML 2.0 speech/IVR media
- VoiceXML 2.1 extensions
- CCXML 1.0 call control
- SRGS 1.0 speech grammars
- SSML 1.0 speech markup
- SISR speech semantic interpretation
- Extensible Markup Language (XML) 1.1
- Namespaces in XML 1.1
- XML Document Object Model
- XML Path Language (XPath) 1.0
- XML Event Syntax
- SOAP Web Services
- WSDL Web Service Description

De Facto Standards Support

- Nuance GSL grammar format
- Scansoft/Speechworks grammar format
- Java speech grammar format (JSGF)

Sample Phone/VoIP/Speech Apps

- Plain Old Interactive Voice Response
- Speech-driven Voice Response
- Intelligent VOIP call-routing applications
- Value-added content services for telecom providers
- IP-PBX solutions
- IP-call center solutions
- Conferencing solutions
- Call recording solutions
- Information retrieval
- Telephone surveys
- Emergency notifications
- Virtual receptionists
- Call center automation
- Order status
- Package tracking
- Conferencing solutions
- Audiotext platforms
- Predictive dialers
- Voicemail services
- Dating chat services
- Voice instant messaging
- Voice verification