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Alvaria Prophecy

Alvaria Prophecy is an industry-leading IVR and SIP platform that makes it easy for companies to interact with people in ways that improve service, drive sales, and lower costs. Built on the core principle of simplicity, Alvaria Prophecy rigorously adheres to open standards. We believe that an IVR and SIP platform should be easy to install, manage, and develop on – and should never suffer at the expense of encumbered business models and inflexible processes.

ADVANTAGES OF ALVARIA PROPHECY INCLUDE:

Foundation for Omnichannel Customer Service

Using Alvaria Prophecy in combination with Alvaria CXP, the industry leading application lifecycle management platform for IVR and digital self-services, allows you to multiply your ROI by moving beyond voice telephony interactions and offering a true omnichannel customer experience on channels such as SMS, mobile web, social chat and many other type of integrations as professional services engagements.

Reduced Complexity of Application Development

Alvaria addresses a wide range of developer skill sets and preferences so you can deploy better applications faster and make changes with ease. Write Alvaria Prophecy applications directly in VoiceXML (for IVR) and CCXML (for advanced call control); or use Alvaria CXP Pro, to benefit from a rich development environment that includes a drag and drop graphical user interface for the design, development, deployment, and reporting of VXML, SMS and Web applications.

Real-time Application Control with Commander

See your application stats in real time, provision network resources on the fly, configure and manage multiple distributed systems, and more, with Alvaria Prophecy Commander, a web-based administration interface.

Better, Smarter Service with Logging

Fine-tune your applications to improve service and lower costs. Alvaria Prophecy collects and indexes data and call logs in real time. Logs can be used with third party logging dashboard providers or BI tools. We also offer a bundle with Alvaria CXP (Alvaria CXP Pro) that adds out-of-the-box integration with popular business intelligence tools and deep analytics covering business metrics such as task completion rates, personalization statistics, no input/no match statistics, caller loyalty, and more.

Native SIP Support

With over 15 years of native SIP support Alvaria Prophecy delivers a clean, scalable SIP foundation that brings together diverse applications and devices, eases access to enterprise data, and lowers the cost of transferring calls with direct IP to IP connectivity. Alvaria Prophecy is fully SIP compliant and IMS ready. Alvaria brings over one and a half decades of experience with worldwide large-scale deployments of SIP solutions.

Deployment Options

Alvaria offers cloud and on-premise deployment options based on a common platform. Customers can easily implement a hybrid solution or move from one solution to the other without having to rewrite their applications. Likewise, Alvaria won't lock you in by delivering proprietary tools on top of the open VoiceXML standard.

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Flexible Pricing

Alvaria Prophecy has built-in capabilities for flexible pricing models that aren't typically available with premise solutions. Alvaria offers a port-lease option and per-minute pricing that enables you to deploy the Alvaria Prophecy platform on your premise with pay-as-you-go billing and no upfront costs.

Alvaria Prophecy 22 Highlights

- 100% VoiceXML 2.1-compliant
- World's most proven and widely-used CCXML engine
- Built-in high-quality speech synthesis
- HD Audio
- Migration to IPv6
- Integrates with third party speech engines via MRCP v1 and v2
- Built-in conferencing
- Built-in call recording
- Supports both voice (VoIP) and fax (FoIP) services
- Built for SIP from the ground up

MORE ALVARIA PROPHECY FEATURES AND CAPABILITIES

Operability

Full-featured, web services-based Provisioning API enables Alvaria Prophecy operations and management to be fully automated; for example: configuration of servers and virtual platforms, provisioning of ASR (speech recognition) and TTS (speech synthesis) resources, provisioning phone numbers and applications, and managing users.

Virtualization

To save operational costs and simplify maintenance, the full Alvaria Prophecy stack can run inside of virtualized environments, from the administration interfaces down to the media server. Alvaria Prophecy supports VMware ESXi and Microsoft Hyper-V.

Speech Recognition

Broad support for speech recognition (ASR) and speech synthesis (TTS), enabling mixed deployments and easy, automatic migration between underlying engines. Alvaria Prophecy provides MRCP standards-based connections to LumenVox, Nuance, and other engines for speech solutions spanning almost 100 languages and dialects.

Call Progress Analysis

No more dead air when picking up a call from an automated system. Alvaria delivers industry-leading call progress analysis (CPA) capabilities for outbound IVR. CPA offers advanced detection of humans versus answering machines, fax machines, and other special information. Alvaria also allows customers to define the business logic associated with the different tones for improved handling and a better customer experience.

Secure, Two-way Call Recording Platform

Record any IVR or agent call without the expense of a separate recording platform. Alvaria Prophecy supports SRTP (encrypted media/audio streaming) and SIP/TLS (a secure version of the SIP signaling protocol). This allows for Alvaria Prophecy to be used in highly secure environments where all aspects of VoIP traffic must be encrypted. The ability to record call conferences is also supported.

HD/Wideband Audio

Traditional phone networks transmit a narrow range of sound frequencies resulting in the "phone" sound we have become accustomed to. With IP communications you can now use HD audio, also called "Wideband" to establish superior sounding phone calls to applications and contact centers. HD Audio gives the caller the feeling of "being there" with others on the call. Newer wideband and narrowband Media Codecs include ISAC (Internet Speech Audio Codec), ILBC (Internet Low Bitrate Codec), and the highly versatile Opus codec, which covers anything from narrowband to fullband.

Security

New security feature to detect and reduce impact of potential Denial of Service attack via SIP. Detection of abnormal activity is attempted on the transport layer (TCP), SIP layer, and CCXML application layer.

Distributed Conference Manager

Alvaria Prophecy's VoiceXML and CCXML-based, speech-driven conference manager features phone and web-based conference call creation, access, and management. Alvaria Prophecy supports conferencing for up to 300 participants with built- in echo cancellation and noise suppression, dual band automatic gain control (agc), and the ability to add, remove, and mute participants.

Conferences can now be recorded in CCXML, you can now initiate recordings, including multiple recordings of the same conference. The recording can transcend the session it was started from. It is possible to pause, resume and stop recordings, and to terminate the recording when the conference is terminated.

CTI Support

To support contact center environments, Alvaria Prophecy integrates with CTI interfaces from Aspect® Unified IP®, Alvaria Cloud, Cisco ICM, Genesys T-Server, Avaya, and Nortel. VoiceXML and CCXML application triggers or events can be used to initiate CTI events and CTI events can be used to trigger VoiceXML call session events. Additionally, being natively based on the SIP protocol, Alvaria Prophecy can support the transfer of CTI information in SIP header messages to integrate with next-generation and IP communications-based environments.

Platform Flexibility

Alvaria Prophecy works with Windows Server 2012, 2016, 2019 and Linux (CentOS or RedHat Enterprise Linux 6.x - 7). Alvaria Prophecy is also available as a SaaS solution, powered by the Alvaria Public Cloud.

Full-featured Configuration and Provisioning API

The full configuration and provisioning functionality is also available a versatile, secure, web service-based API. This includes user managements, server and platform configuration, application and number provisioning, and more. With this API, service providers can automate routine tasks, integrate the Alvaria Prophecy platform into their management toolset, and/or create custom management GUIs to their end customers.

Sample Applications

- DTMF-driven Interactive Voice Response
- Speech-driven Voice Response
- Intelligent VoIP call routing applications
- IP-PBX solutions
- IP call center solutions
- Call recording solutions
- Information retrieval

- Telephone surveys
- Outage notifications
- Virtual receptionists
- Order status
- Conferencing solutions
- Voice verification

Carrier-grade Performance and Scalability

Alvaria Prophecy scales from deployment on a developer notebook to server clusters spanning multiple data centers, supporting 10s of thousands of telephony ports. All Alvaria Prophecy components communicate via SIP, RTP, and MRCP protocols, making it easy to distribute any functions onto additional servers. Customers can easily increase core platform capacity at any time simply by adding gateway and server components to the existing system. Call routing at each tier enables seamless failure detection with the ability to follow multiple call paths for no single point of failure.

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TECHNICAL SPECIFICATIONS

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 authentication support
- Configurable SIP port range
- NAT IP translation support

SIP Compatibility

- SIP from: Global Crossing, Level (3)
- Verizon, AT&T, Avaya, Cisco, BT, Voxbone, Polycom, Sipura, Sonus, Ribbon
- Works with most other SIP solutions

Supported Audio Codecs

- PCMA
- PCMU
- G.722
- G.726-32
- GSM
- L16
- G.729
- Speex
- AMR (NB/WB)
- iSAC
- iLBC
- OPUS

Media and IVR Features

- Audio prompt/announcement playback
- Audio bridging
- Audio mixing/conferencing
- Audio noise removal
- Audio fixed gain control
- Audio call recording
- Automated Speech Recognition (ASR)
- Text-to-speech (TTS)
- Audio playout jitter buffer
- DTMF tone detection and generation
- Outbound calling
- Intelligent Call Progress Analysis (CPA)
- Supports VoiceXML 2.x IVR
- Runs on standard x86 platforms
- Inbound/Outbound Faxing
- Wideband Audio

W3C Standards Support

- VoiceXML 2.0/2.1 speech/ IVR media
- CCXML 1.0 call control
- SCXML 1.0 (draft) flow 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
- JSR 154 Java Servlet 2.5
- JSR 254 Java Server Pages 2.1

De Facto Standards Support

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

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IETF Standards Support

- RFC 3261 SIP
- RFC 3310 SIP authentication
- RFC 5630 SIP TLS
- RFC 3262 Reliable provisional SIP Responses
- RFC 3265 SIP - Event Notification
- RFC 4032 Update to SIP Preconditions Framework (*)
- RFC 3515 SIP Refer Method (*)
- RFC 3311 SIP UPDATE Method
- RFC 6086 SIP INFO Method
- RFC 3372 SIP-T: Context and Architectures (*)
- RFC 4566 SDP
- RFC 3264 SDP negotiation
- RFC 4568 SDP Security Descriptions
- RFC 3550 RTP
- RFC 3551 RTP Profile for Audio
- RFC 3711 Secure RTP (SRTP)
- 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
- RFC 5552 SIP interface to VoiceXML media servers
- RFC 4612 Real-Time Fax (T.38)
  (*) = Partial support

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