WHITE PAPER

Aspect® Advanced Voice Portal (AVP)

Aspect Advanced Voice Portal (AVP) is an industry-leading communications platform that makes it easy for companies to interact with people in ways that improve service, drive sales, and lower costs. Take advantage of AVP to deliver everything from speech-driven self-service and outbound interactive voice response (IVR) messaging, to unified communications and innovative SIP applications.

Built on the core principle of simplicity, AVP rigorously adheres to open standards. We believe that a voice application 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 AVP include:

Unified Self-Service™. Using AVP in combination with Aspect CXP allows you to multiply your ROI by leveraging voice interactions plus SMS, Twitter, web-chat, mobile web applications and Smartphone apps, for a true omni-channel customer experience.

Reduced complexity of application development. AVP addresses a wide range of developer skill sets and preferences so you can deploy better applications faster and make changes with ease. Write AVP applications directly in VoiceXML and CCXML; or use CXP Pro, the bundle of Aspect CXP and AVP, to benefit from a rich development environment for the design, development, deployment, and reporting of IVR 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.

Better, smarter service with logging and analytics. Fine-tune your applications to improve service and lower costs. AVP collects and indexes data and call logs in real time, giving IT staff and developers the ability to search, navigate, analyze, measure, and report statistics, transactions, and errors as they occur. The CXP Pro bundle adds out-of-the-box integration with popular business intelligence tools and deep analytics covering business metrics such as task completion rates, personalization statistics, caller loyalty, and more.

Native SIP support. AVP 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. AVP is fully SIP compliant and IMS ready. Aspect brings over a decade of experience with worldwide deployment of SIP solutions.

Deployment options. Aspect offers on-demand hosting 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, Aspect won't lock you in by delivering proprietary tools on top of the open VoiceXML standard.

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

AVP Highlights

- 100% VoiceXML-compliant
- World's most proven and widely-used CCXML engine
- Built-in high-quality speech synthesis
- HD Audio Migration to IPv6
- Easily works with third party speech engines via MRCP
- Built-in conferencing
- Built-in call recording
- Supports both voice (VoIP) and fax (FoIP) services

- Built for SIP from the ground up
- Ready for deployment in PCI-DSS Level 1-compliant data centers

More Aspect Advanced Voice Portal Features and Capabilities

Operability

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

Virtualization

To save operational costs and simplify maintenance, the full AVP stack can run inside of virtualized environments, from the administration interfaces down to the media server. AVP 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. AVP provides MRCP standards-based connections to LumenVox, Nuance, and other engines for speech solutions spanning 52 languages. In addition, Aspect bundles its own highly accurate speech synthesis engines.

Call Progress Analysis

No more dead air when picking up a call from an automated system. Aspect 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. Aspect 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. AVP provides Payment Card Industry (PCI) compliant call recording via its support for public-key encrypted audio files. AVP supports SRTP (encrypted media/audio streaming) and SIP/TLS (a secure version of the SIP signaling protocol). This allows for AVP to be used in highly secure environments where all aspects of VoIP traffic must be encrypted.

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. New 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.

Distributed Conference Manager

AVP's VoiceXML and CCXML-based, speech-driven conference manager features phone and web-based conference call creation, access, and management. AVP 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.

CTI Support

To support contact center environments, AVP integrates with CTI interfaces from Aspect® Unified IP®, 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, AVP can support the transfer of CTI information in SIP header messages to integrate with next-generation and IP communications-based environments

Aspect® CXP

CXP Pro edition provides all the power of AVP bundled with the Aspect CXP Application Lifecycle Management suite. Aspect CXP provides real-time reporting and deep analysis of caller behavior, application performance, and transaction success for rapid tuning, improved ROI, and a better customer experience. It also facilitates a consistent, personalized user interface across all customer service touchpoints.

Platform Flexibility

AVP works with Windows Server 2012 and Linux (Centos 6 or Redhat Enterprise). AVP is also available as a pre-installed, pre-configured turnkey server.

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 AVP platform into their management toolset, and/or create custom management GUIs to their end customers.

Aspect AVP Advantages

- Take advantage of the latest standards to simplify development, speed deployment, and ease ongoing maintenance
- Leverage easy-to-use development tools and resources
- Overcome barriers to speech adoption and improve the customer experience with AVP's free recognition and synthesis engines; or use any MRCP-compliant speech engine
- Deploy blended inbound and outbound applications
- Enable customers to contact you in more ways – and save money – with Unified Self-Service™
- Remove the limitations of expensive legacy systems and eliminate the need for proprietary skill sets
- Leverage existing investments in web infrastructure and applications, including back end integrations and business logic
- Centralize management and reporting through web-based tools
- Deploy in front of or behind the PBX and in both TDM and IP environments
- Native IPv6 Support
- Deploy on a single server or in a distributed configuration
- Realize carrier-grade reliability and management of sophisticated, multi-tenant environments
- Leverage off-the-shelf hardware
- Realize a seamless migration path to a next-generation VoIP network
- Get 24x7x365 support from Aspect Customer Care, including our knowledgeable, highly-trained customer engineers
- Supports the latest Nuance and LumenVox ASR engines for advanced speech recognition

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

AVP scales from deployment on a developer notebook to 1000+ server clusters spanning multiple data centers. All AVP 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.

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 registrar server support
- SIP proxy server support
- SIP authentication support
- Configurable SIP port range
- NAT IP translation support
- Multi-ethernet card bridging support

SIP Compatibility

- SIP from: Global Crossing, Level (3),
- Verizon/ MCI, AT&T, Avaya, Cisco, Nortel, BT, Voxbone, Polycom, Sipura, Sonus
- 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 dynamic 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 289 SIP Servlet 1.1
- 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)

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

Corporate Headquarters East
300 Apollo Drive
Chelmsford, MA 01824
+(1) 978 250 7900 office
+(1) 978 244 7410 fax

Corporate Headquarters West
2325 E. Camelback Road,
Suite 700
Phoenix, AZ 85016
+(1) 602 282 1500 office
+(1) 602 956 2294 fax

Europe & Africa Headquarters
2 The Square, Stockley Park
Uxbridge
Middlesex UB11 1AD
+(44) 20 8589 1000 office
+(44) 20 8589 1001 fax

Asia Pacific & Middle East Headquarters
8 Cross Street
25-01/02 PWC Building
Singapore 048424
+(65) 6590 0388 office
+(65) 6324 1003 fax

About Aspect

Aspect's fully-integrated solution unifies the three most important facets of modern customer engagement strategy: customer interaction management, workforce optimization, and back-office. Through a full suite of cloud, hosted and hybrid deployment options, we help the world's most demanding contact centers and back offices seamlessly align their people, processes and touch points to deliver remarkable customer experiences. For more information, visit **www.aspect.com**.

aspect™