

# voxpilot Open Media Platform 3.2

## Technical Data Sheet

### Overview

Voxpilot Open Media Platform (OMP) is a distributed call control and VoiceXML-based media processing platform designed to enable rapid delivery of next generation interactive DTMF, speech, and multimedia services on a single platform. Employing a unique combination of advanced PSTN features with VoIP capabilities, the Voxpilot Open Media Platform is optimised both for the networks of today and also those of tomorrow.

The OMP contains the Voxpilot VoiceXML Media Server and Voxpilot Call Server. Voxpilot's open and modular architecture allows both components to be deployed either together or independently thus delivering both a complete platform solution and full-featured building blocks for next generation networks.

Whether your aim is to offer high quality customer self-service, reduce operational costs, or generate new revenue streams, the Voxpilot Open Media Platform delivers proven robustness, flexibility, and scalability at a reasonable price.

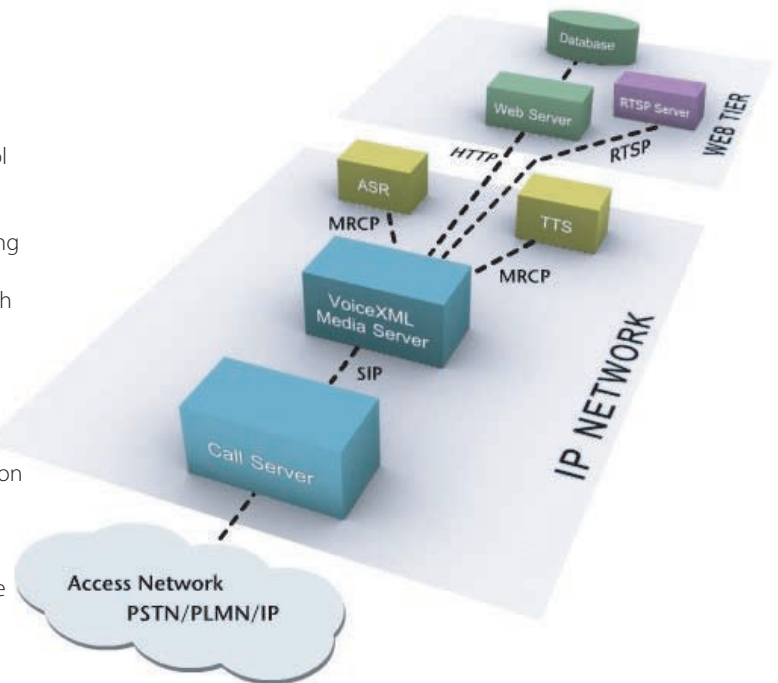
### Key Features

#### Open Standards and the Web Model

The Voxpilot OMP architecture leverages W3C and IETF open standards, which revolutionize the way in which voice services are deployed, bringing IVR from the old world of proprietary platforms into the Internet age. These standards include VoiceXML 2.1 compliance as well as support for SSML, SRGS, SISR and CCXML. Voxpilot employs the Media Resource Control Protocol (MRCP) for integrating leading media resources such as speech recognition and speech synthesis engines.

VoiceXML introduces the Web application development and deployment model to the world of telephony-based interactive services. Existing investments in Web infrastructure such as Web servers, application server-based business logic, and databases can be leveraged to facilitate rapid development of new and innovative applications and services. Moreover, the Web model allows companies to exploit their existing Web expertise to build Interactive Voice and Video Response (IVVR) solutions and apply them to revenue generating telephony and VoIP networks.

## Next Generation Interactive Voice & Video Services



### IP Convergence

The unique combination of VoIP and PSTN interfaces supported by the Voxpilot OMP makes it ideally positioned to support the migration from traditional telephony networks to next generation "all-IP" network architectures such as the IP Multimedia Subsystem (IMS).

The Third Generation Partnership Project (3GPP/3GPP2) introduced the IMS architecture as part of their vision to drive convergence of the cellular and Internet worlds by seamlessly integrating voice, data, and multimedia services on the same network. Fueled largely by the fact that IMS is access network independent and hence ideal for wireless-wireline convergence, it is fast becoming the leading reference architecture for next generation networks. It will allow deployment of interactive TV, streaming of multimedia files, and IVVR services on any IMS compliant network thereby safeguarding platform investment.

IMS leverages open standard IETF protocols with SIP notably providing the core signalling protocol. Through SIP, the Voxpilot OMP integrates easily into IMS network architectures providing a rich media resource function (MRF) feature set to SIP Application Servers. With over 6 years experience developing and deploying VoIP technologies, Voxpilot is committed to supporting next generation all-IP network architectures.

# More powerful, More open, More flexible.....

## Speech Engine Agnostic

The Voxpilot OMP supports the best-of-breed speech recognition and speech synthesis engines. Customers have the flexibility to choose their preferred speech engine vendor based on their specific requirements for language support and features. Engines from different vendors may even be configured in the same deployment. With the advent of open protocols such as MRCP, customers avoid "vendor lock-in".

## Carrier-Grade Quality

Voxpilot's platform technology has had many years of deployments executing applications written by thousands of developers. The result is a robust, carrier-grade product with proven scalability. Employing SIP, MRCP, HTTP, and SNMP protocols in the platform design facilitates scalable, load-balanced, and maintainable clusters of Call Server, VoiceXML Media Server, and speech engine resources to be deployed in a reliable fashion.



As an active member of the standards community, Voxpilot was one of the first companies to achieve VoiceXML Forum Platform Certification, which involved independent and rigorous testing of all VoiceXML 2.0 features. Voxpilot is both a contributor and editor of W3C and IETF specifications and committed to the development and evolution of

open standards. It also promotes advancements like video processing with VoiceXML to be part of the upcoming VoiceXML 3.0 standard.

The Voxpilot OMP leverages the leading RADVISION SIP stack to provide high quality, interoperable, and scalable SIP connectivity to the Call Server and VoiceXML Media Server. It supports Red Hat Enterprise Linux and Microsoft Windows operating systems and runs on standard Intel hardware. The platform also addresses both low-end and high-end densities, supporting configurations ranging from 2 ports up to 480 ports per server.

## Engineered for Video

The availability of new media features, driven by improved terminal and network capabilities, is enabling the delivery of new and exciting video applications and multimedia services. In addition to supporting rich audio processing capabilities such as speech synthesis and speech recognition, the Voxpilot Open Media Platform is engineered to support video applications and multimedia services both in currently deployed 3G networks (3G-324M) and directly in IP-based network architectures such as IMS.

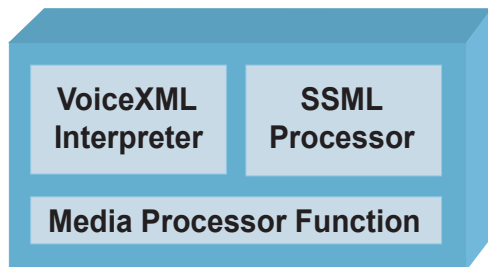


VoiceXML, while originally developed for speech and DTMF interactive services, is a natural paradigm for delivering rich, video-enabled services. Video is simply another media channel in a human-machine dialog and, to that end, standard functionality such as video playback, record, and data push to web servers is possible today through standard VoiceXML. The combination of rich audio and video capabilities, offered through a single Web-based service creation approach such as VoiceXML, results in an exceptionally powerful multimedia service delivery option that goes far beyond what has been available up to now.

## Multi-Application Platform

The Voxpilot Open Media Platform can support a wide range of VoiceXML powered self-service applications including:

- ▶ Interactive Voice and Video Response Services
- ▶ Telecom Service Platform and Intelligent Routing
- ▶ Conferencing and Voice/Video Mail Solutions
- ▶ Multimedia and Multimodal services
- ▶ Call Centre Self-Service and Automated Directory Assistance
- ▶ Voice/Video Service Outsourcing/Hosting
- ▶ Voice/Video portal



**Voxpilot VoiceXML Media Server**

The SIP-based VoiceXML Media Server provides a certified VoiceXML 2.0 and VoiceXML 2.1 compliant interpreter and SSML 1.0 processor complete with the requisite host-based media processing functions including audio/video announcement and recording, DTMF detection and generation (RFC 2833), SRGS DTMF recognition, and media forking. It employs MRCP to provide seamless control of speech media resources for performing speech recognition and speech synthesis actions.

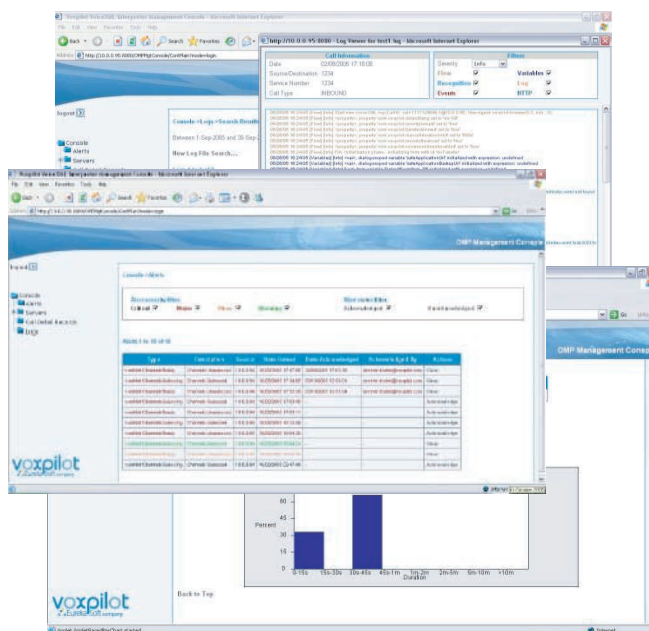
The VoiceXML Media Server allows developers to create advanced interactive video services using standard VoiceXML by leveraging the 3GP multimedia file container format and content from RTSP streaming servers. It can interoperate directly with the Call Server for 3G-324M services or with any standard SIP-based video gateway.

The Voxpilot VoiceXML Interpreter is implemented natively for optimum performance, and has been designed to be highly robust and scalable and includes an active health monitoring system for detecting and alerting to potential problems. The Voxpilot SSML Processor supports all leading TTS engines and the Voxpilot AudioEngine for streaming, caching and transcoding local and remote audio files.

The VoiceXML Media Server includes numerous configuration parameters to allow for effective customisation and extension modules can be easily plugged in for interfacing directly with external systems. Real-time control and monitoring is achieved through Web consoles.

In traditional PSTN or PLMN networks, the Call Server can interface within the Intelligent Network (IN) as an Intelligent Peripheral (IP) or a Service Control Point (SCP) via the INAP and CAMEL protocols. The Call Server simultaneously addresses next-generation telephony requirements by fully supporting the SIP/RTP protocols for VoIP and providing signalling and media gateway functionality between IP and circuit-switched networks.

The Call Server may be programmed via CCXML or Orateur scripts (an event-driven programming language). Extensions and libraries may be created and added to the Orateur language to interface with external systems. The Call Server is integrated with industry-leading CTI environments by means of specific extension packages. For optimum performance, the Call Server employs hardware-based DSPs for media processing and signalling.



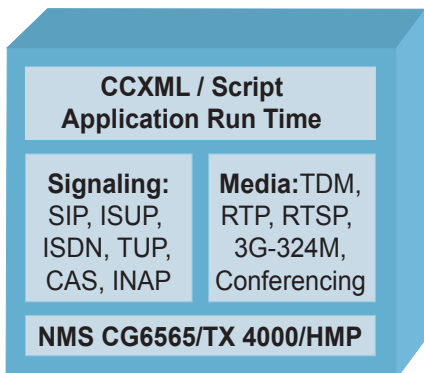
**OMP Management Console**

The OMP Management Console is a powerful, flexible, and tailored management station, which delivers a centralized view of a live OMP platform to both system administrators and business managers. The OMP Management Console is an optional component ideal for deployment in network operation centers or the enterprise.

The OMP Management Console offers an extensive list of management features, including:

- Alarm notification and management
- Process control, monitoring, and automatic restart
- VoiceXML application configuration
- Call detail record access and statistics
- Resource allocation and pooling
- Application log search engine and viewer

The OMP Management Console utilizes industry standard protocols such as SNMP and runs on BEA WebLogic and Apache Tomcat application servers and Oracle and MySQL databases.



**Voxpilot Call Server**

The Voxpilot Call Server provides rich programmatic call control and gateway features to the Voxpilot Open Media Platform. It supports SS7 call control protocols such as ISUP and TUP in addition to IN protocols such as INAP, MAP and CAMEL. The Call Server also readily supports other telecommunications protocols including ISDN, CAS (R2), and 3G-324M video.

## Technical Specifications

### VoIP Interface

RFC 3261 (SIP core)  
RFC 3263 (Locating SIP servers)  
RFC 3264 (Offer/answer)  
RFC 3515 (REFER)  
RFC 2327 (SDP)  
RFC 3550 / 3551 (RTP)  
RFC 2326 (RTSP)  
I-D draft-burke-vxml

### PSTN Interface

ISUP  
TUP  
ISDN  
CAS (R2)  
INAP  
MAP and CAMEL

### Web Standards

VoiceXML 2.0 (certified)  
VoiceXML 2.1  
SRGS 1.0  
SSML 1.0  
SISR 1.0  
CCXML 1.0  
HTTP(S) 1.1

### Media Gateways

Alcatel OmniPCX 4400 (IP PBX)  
Asterisk PBX  
AudioCodes Mediant Gateways  
AudioCodes TP-260  
Cisco AS5xxx  
Dialogic TIMG3000DTI  
Dilithium DTG-2000 (3G-324M)  
Pika Gateway  
Radvision N30 3G Gateways  
and others

### IMS Functions

MRF- Media Resource Function  
MRFC & MRFP

### Video Standards

3G-324M  
H.263, AMR  
MPEG-4 Part 10 (AVC): H.264<sup>1</sup>  
3GP File Format  
Video support in VoiceXML  
IP Interface:  
RFC 2190 / 2429 (H.263)  
RFC 3016 (MPEG-4)  
RFC 3267 / IF2 (AMR)

### Media Processing

G.711, G.723.1, G.726, G.729a/b  
HTTP streamed audio  
RTSP Audio and/or Video Streaming  
Voice Activity Detection  
Echo Cancellation  
Conferencing  
DTMF Recognition and Generation  
(RFC 2833, SRGS)  
Text/Image/Audio Overlay  
VCR Controls

### Streaming Servers

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

### APIs

Call Control Scripting Language  
VoiceXML <object> API  
Authorization and Usage API  
Outbound Call Placer API

<sup>1</sup> Available Q1 2008

### OA&M

SNMPv2c / SNMPv3  
Centralized Management Console  
Health Monitor  
Call Detail Records (CDR)  
Instrumentation and Logging  
DNIS to URI Mapping  
Process Watchdog

### Speech Recognition

Loquendo ASR 7.2  
Lumenvox  
Nuance v8.5, v9  
Nuance OSR 3  
Telisma TeliSpeech 1.2  
Vocative v1.0  
MRCPv1 compatible engines

### Speech Synthesis

Acapela Sayso 5.1  
CVOXv1  
Loquendo 6.5  
Nuance Realspeak 4  
Nuance Rhetorical 4.2  
MRCPv1 compatible engines

### CTI

Genesys T-Server  
CosmoCom CosmoCall Universe  
Kiamo

### Telephony Components (Call Server)

NMS CG 6000  
NMS CG 6565  
NMS TX 4000  
NMS Host Media Processing (HMP)

### Operating Systems

Red Hat Enterprise Linux ES 3  
Microsoft Windows Server 2003

## About Voxpilot

Voxpilot brings the next generation of interactive telecommunication services to enterprises, integrators, carriers and service providers. Our flagship product, Voxpilot Open Media Platform (OMP), has been installed in a variety of networks across Europe, America, Africa and Asia. Combining advanced telephony features with Internet capabilities such as SIP and Web technologies such as VoiceXML, the Open Media Platform replaces classic Interactive Voice Response and enables speech-based self-service, conferencing, multimedia services, and video-based solutions such as VideoMail. Voxpilot's customers include France Telecom, Wind, Cegetel, Monaco Telecom, BT, NMS Communications, Swisscom, Ericsson, Cofroute, Skype, PostFinance (Swiss Post Telephone Bank), and Universal Music Mobile.

For more information on our products and services, please contact us at the numbers below or [sales@voxpilot.com](mailto:sales@voxpilot.com) and visit our web site at <http://www.voxpilot.com>.

#### France:

4, Avenue des Martyrs de  
Soweto,  
91300 Massy  
France.  
Tel: +33 1 69 75 78 00  
Fax: +33 1 69 75 78 01

#### Ireland:

6-9 Trinity Street,  
Dublin 2,  
Ireland  
Tel: +353 1 209 1919  
Fax: +353 1 209 1920

#### USA:

8 Cog Hill Drive  
Simpsonville, SC 29681  
USA  
Tel: +1 864 627 1442  
Fax: +1 864 627 1222

#### Germany, Austria, Switzerland, Italy:

c/o Plurinnova,  
Wetlistrasse 2,  
8032 Zürich,  
Switzerland.  
Tel/Fax: +41 43 534 91 63