TELESOFT

DATASHEET

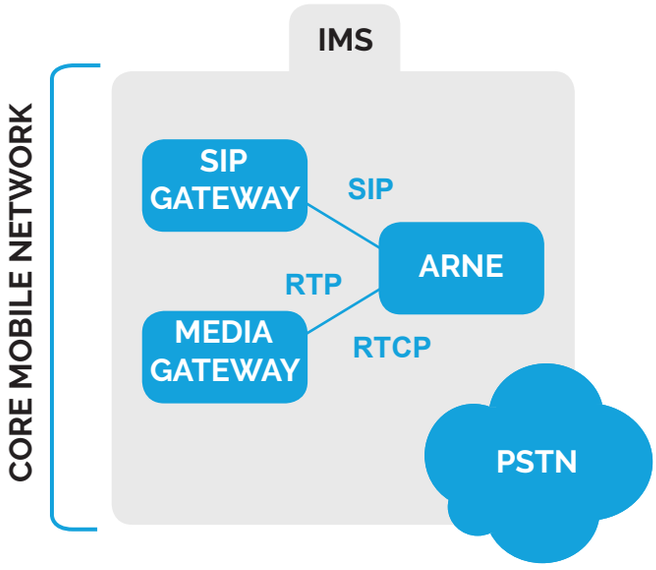
vARNE IVR

The vARNE IVR is a software based IVR that processes up to 2000 channels of SIP & RTP per 12 cores (24 threads), allowing complete system optimisation and flexibility.

The Telesoft vARNE IVR is a software based optimised system specifically developed for telecom value-added services and enterprise/call-centre applications. Allowing operators, OEMs, and system integrators to quickly deploy advanced interactive voice services in fixed, cellular, and next-generation telecom networks.

Scalable in intervals of up to 2000 channels per 12 cores (24 threads) in a virtualised environment, the vARNE can grow with your needs, minimising investment, and maximising return. Multiple codec support enables the operator to manage network bandwidth efficiently. Telesoft's field-proven SIP and SIP-I interfaces enable trouble-free integration of your applications to the network, ensuring new services get to market quickly.

Built using open standards, the vARNE IVR runs your voiceXML and call control XML (CCXML) compliant applications. Both are XML-based markup languages commonly used in IVR deployments, allowing fast development and modification of applications.



KEY FEATURES	
Up to 2000 channels in a single system	Lower cost per channel
Multiple, different services implemented as different CCXML or VXML scripts can be run at the same time allowing a single IVR instance to run many different services	Enables specific CCXML or VXML scripts (and hence services) to be invoked based on OLI and TLI
Field proven VoiceXML and CCXML	Reuse existing applications and quickly deploy new services
Multiple Protocol Support (SIP/SIP-I)	Suitable for network migration
Supports multiple codecs including AMR and Transcoding between narrowband and wideband	Reduced bandwidth and network overheads
Run in a virtualised environment	Use on existing infrastructure
MRCP Support	Flexibility to create bespoke configuration based on requirement
Fully featured User Interface	Message counts and ratios, per-port status, per-service statistics and system alarms are displayed in real time
Enhanced statistics	Tabulate and graph historical statistics including signalling, call control, and per-service statistics. Statistics are collected on a per-minute basis over several months. Observe and compare trends for different hours, days, weeks, or months

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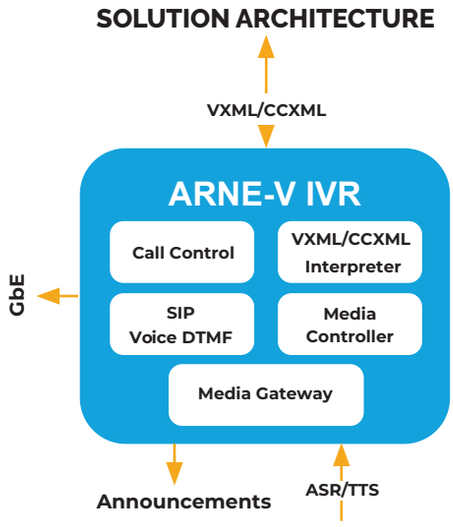
SOLUTION ARCHITECTURE



APPLICATIONS	
Automated customer care	Ring back tones
Voice mail	Single IVR (Multi-tenanted) deployments
Billing	Outbound dialling
Infotainments services	Suitable for network migration

TECHNICAL SPECIFICATIONS	
Network Interface	• Gbit Ethernet SIP/RTP interface • RTP: IETF RFC3550 • IPv4, IPv61l
VXML/CCXML	• VoiceXML: W3C voice extensible markup language v2.0 • VoiceXML sessions on all calls simultaneously • CCXML: W3C call control markup language V1.0
Media Processing	• Up to 2,000 channels • Voice/announcement playback • RFC2833 DTMF • Built-in grammars for dates/times/currencies • file://access to local media files • http(s)://access to remote media files • Content caching for improved latency • Up to 32 languages • Base platform supporting audio G.711 codec. *Optional support AMR (3GPP TS 26.071 (RFC4867)), AMR-WB (3GPP TS 26.171), ITU-T G.729 06/2012
Management	• Simple configuration and resource management • Telnet and FTP for remote debug and download • Alarms and statistics via SNMP
Control Interface	• SIP: IETF RFC3261 • SIP-netann: RFC 4240 • RTP Payload for DTMF Digits: RFC2833
Virtualisation Platform Support	• OpenStack • Promox Virtual Environment • XEN Project Virtual Machine • KVM/QEMU • VMWare ESXi • Amazon Web Services (AWS) • Microsoft Azure *separate chargable license applies

VoiceXML Forum Certified
The ARNE IVR incorporates a VoiceXML 2.0 interpreter certified by the VoiceXML forum. VoiceXML (VXML) is the W3C's standard for specifying interactive voice dialogues between a human and a computer.

CCXML
The ARNE IVR implements the W3C Call Control Extensible Markup Language V1.0. CCXML is an event driven language designed to be compliant with VXML dialog control and allows applications to manipulate call legs.

Single IVR (Multi-tenanting)
The ARNE supports a multi-tenant business model, enabling multiple "tenant" organisations to execute services on a single, shared ARNE IVR made available through a single or multiple hosting organisation(s).

DTMF User Interaction
The ARNE IVR allows DTMF to be detected on every channel simultaneously, received as RFC2833 RTP Payload for DTMF Digits.

Configuration and Maintenance
The ARNE IVR supports a simple mechanism for configuration, designed to make both installation and maintenance fast and simple, reducing overheads that can occur from both downtime and lengthy installations.

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