Datasheet

ARNE IVR

The ARNE IVR is a complete, integrated IVR that processes up to 2000 channels in a small footprint, low power 1U chassis, reducing interconnect and OPEX costs, power and TCO

The Telesoft ARNE IVR is a hardware 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 and video services in fixed, cellular and next-generation telecom networks.

Scalable from 240 to 2,000 channels in a single unit or via rack and stack approach, the ARNE IVR grows with your needs to minimise investment and maximise return. Using Telesoft's field-proven SIP, ISUP and ISDN interfaces to enable trouble-free integration of your applications to the network ensuring you get new services to market quickly.

Built using open standards, the ARNE 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.

Telesoft has supplied telephony products to the world's leading operators and OEMs for over 20 years. This deep industry knowledge and experience ensures the ARNE IVR platform is designed and built to the highest standards to provide years of reliable service.

Arne IVR - STM-1/16E1/8E1/2 x 1GbE

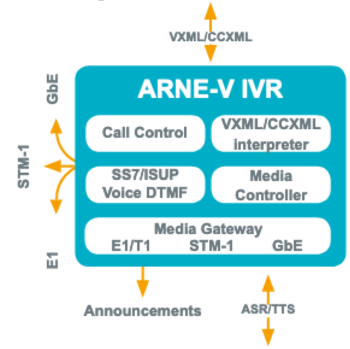
Key Features

Up to 2000 channels in a single system	Lower cost per channel
Field proven VoiceXML and CCXML	Reuse existing applications and quickly deploy new services
Small footprint 1U Telecoms Server	Save power and save space
Multiple Protocol Support (SIP/SIP-I/ISUP/ SIGTRAN)	Suitable for network migration
Integrated application server option	Combine IVR and app server in a single 1U chassis
Redundant hardware options	Protects from hardware failures
Connects to optical STM-1	Consolidate multiple E1s into a single optical STM-1
MRCP Support	Text-to-speech and automatic speech recognition supported

Applications

Automated customer care	Ring back tones
Voice mail	Multi-tenanted deployments
Billing	Outbound dialling
Infotainments services	

System Architecture



Performance Figures

Variant	Media Channels	BHCA
8E1	240	250,000
16E1	480	500,000
STM-1 (63E1)	2000	500,000

Technical Specifications

Network Interfaces

- 8xE1 TDM interface or 16xE1 TDM interface or STM-1 TDM interface or
- Gbit Ethernet SIP/RTP interface
- RTP: IETF RFC3550
- IPv4, IPv61

VXML/CCXML

- VoiceXML: W3C voice extensible markup language v2.0
- SSML: W3C speech synthesis markup language V1.0
- SRGS: W3C speech recognition grammar specification V1.0
- SISR: W3C semantic interpretation for speech recognition V1.0
- MRCP: IETF media resource control protocol 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
- DTMF and VAD
- 3-way conferencing
- ASR/TTS via MRCP v2 to third party speech engine
- 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
- Multiple language support
- Base platform supporting audio G.711 codec

Management

- Simple configuration and resource management
- Graphical state visualisation Telnet and FTP for remote debug and download
- Alarms and statistics via SNMP

Platform

- 1U 19" chassis and 100-240v AC power
- Temperature: 0°C to 50°C (operating)
- Humidity: 8% to 90% (operating)

Control Interfaces

- SIP: IETF RFC3261
- SIP-netann: RFC 4240
- RTP Payload for DTMF Digits: RFC2833
- ITU-T ISUP: Q.761-Q.764
- ITU-T international ISUP: Q.767
- ETSI ISUP V2: ETS 300-356-(basic services)
- Euro ISDN: ETS 300-102

Order Options

Part Number	Description
500002842	STM-1 ISUP TDM 32 E1
500002843	STM-1 ISUP SIGTRAN 32 E1
500002862	STM-1 ISUP SIGTRAN 63 E1
500002782	8E1 ISUP TDM
500002928	8E1 ISUP SIGTRAN
500002811	16E1 ISUP TDM
500002929	16E1 ISUP SIGTRAN
500002684	Year 3 Extended Warranty H/W & S/W
500003025	On-site engineering service

Feature Definition

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.

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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