ARNE IVR

With lowest price/port in the market and a proven heritage, this IVR platform enables revenue-generating voice and video applications to be easily deployed into either IP or TDM network architectures.

KEY FEATURES

- 240 to 2,000 channels
- Plays voice, video and music
- Pre-configured easy connection to SIP, ISUP or ISDN telecom networks
- Specialized hardware, optimized for telecoms VAS applications
- Field-proven VoiceXML/CCXML interpreter
- Text-to-speech and automatic speech recognition supported
- Competitive pricing

KEY BENEFITS

- Complete scalable IVR platform for VAS and customer care applications
- Designed and built to the highest standards to ensure reliability
- Ideal for operators, OEMs and SI developers
- Supports open VoiceXML/CCXML standards enabling easy application development and preventing vendor lock-in

APPLICATIONS

- Self Care
- Music ringback
- Voice & Video VAS
- Infotainment services
- Charging & Billing

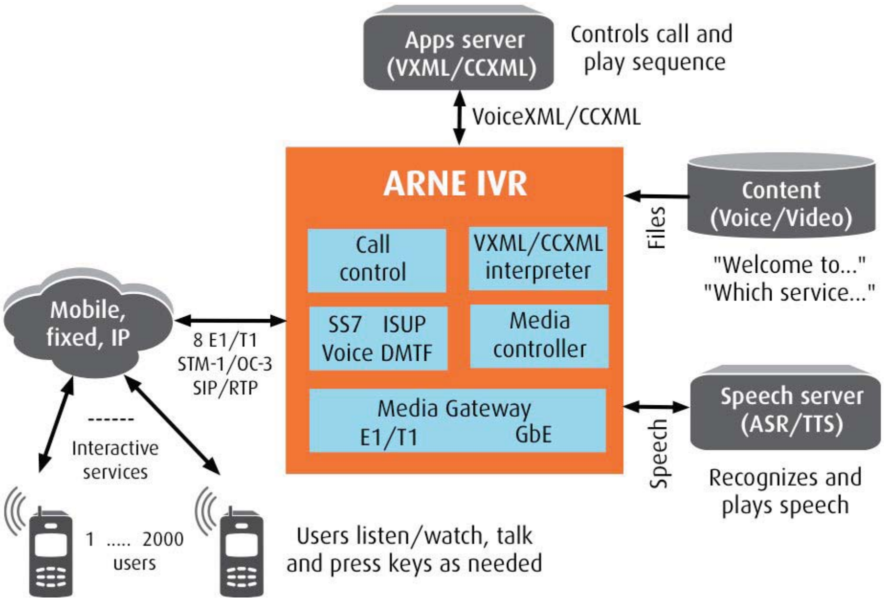
OVERVIEW

The ARNE IVR is specifically optimized for telecom value-added services and enterprise/call-center 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 minimize investment and maximize return. It uses Telesoft Technologies' 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. A low-cost platform, it can be used for automated customer care, infotainment, caller ringback tones, social media and a host of other voice, music and video applications.

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 deployment, allowing fast development and modification of applications. The ARNE IVR supports text-to-speech (TTS) and automated speech recognition (ASR) via an MRCP interface, ensuring a next-generation user experience. Simple configuration makes development and field support easy.

Telesoft Technologies 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.



CERTIFIED VoiceXML FORUM PLATFORM

ARNE IVR

TECHNICAL SPECIFICATIONS

Media Processing and Codecs

- file://access to local media files
- http(s)://access to remote media files
- Content caching for improved latency

Audio

- Codecs:
  - G.711 μ-Law, A-Law
  - AMR, AMR2[1]
  - AAC[1]
- Up to 2,000 channels dependant on platform
- Simultanious voice play on every channel
- DTMF and voice activity detection on every channel simultaneously
- 3-way conference
- ASR/TTS via MRCP v2 to third party speech engine
- Built-in grammars for dates/times/currencies
- Up to 32 languages

Video

- Simultaneous video play on up to 240 channels
- 3GPP file container
- Single format container file or stream play
- Video codecs:
  - H.263, H.263+[1], H.263++[1]
  - MPEG4[1]
- Transcoding, Transrating, Transizing[1]
- Text and Image Overay[1]
- 3GPP 3G-324M
  - ITU-T H.223 Annex A (error handling level 1)
  - ITU-T H.223 Annex B (error handling level 2)
  - ITU-T H.223 Annex C+D (error handling level 3)
  - ITU-T H.245 Version 13 (advanced call control)
  - ITU-T H.324 Annex C (cellular requirements)

Management

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

VoiceXML/CCXML support

- VoiceXML: W3C voice extensible markup language v2.0, v2.1[1]
- 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

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)
- ANSI ISUP: T1.113, Telcordia GR-246-CORE
- UK ISUP: PNO ISC 007
- Indian ISUP: S/CCS-02/03
- Brazilian ISUP: TELEBRAS 22-250-735
- ISDN: Q.931, Belcore TR-NWT-001268, TR-NWT-001187
- AT&T ISDN: TR-41449, TR-41459
- Euro ISDN: ETS 300-102

Network interfaces

- 8 E1/T1 TDM interface (240 channels) or
- STM-1/OC-3 TDM interface (2,000 channels) or
- Gbit ethernet SIP/RTP interface
- RTP: IETF RFC3550
- IPv4, IPv6[1]

Platform

- 1U 19" chassis, 100-240v AC power
- Operating temperature: 0° to 50°C (storage: -40° to 70°C)
- Operating humidity: 8% to 90%
- Linux operating system

[1] Partial implementation, some capability may be in development. Contact sales for availability

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