ARNE IVR
Product Brief

Steve Patton

Where **innovative thinking**
meets **engineering excellence**

telesoft TECHNOLOGIES

ARNE IVR Product Brief

Feature packed, Simple to Use

Introduction

The ARNE IVR is an integrated Interactive Voice and Video response appliance used to deploy telephony based Self-Service applications in Customer Care and Value Added Services environments. Application developers and OEMs use the ARNE to rapidly deploy multi-tenanted CCXML and VoiceXML (VXML) based applications without needing to understand the complex underlying signalling and digital signal processing.

Key Features at Glance:

Connect to Packet Switched (PS) equipment	Connects to converged networks and gateways with BICC and SIP Connects to 3GPP UMTS Rel 4
Connect to legacy Circuit Switched (CS) equipment	Connects to existing wireline and 2G/2.5G wireless networks with SS7-ISUP and ISDN
Supports CCXML 1.0 and VoiceXML 2.0	On-box VXML/CCXML interpreters enable virtually **any service** to be deployed
Outbound dialler	Make outbound calls through CCXML http connector
VCR playback controls	Fast-forward and rewind during prompt playback
Multi-tenanting	Enables specific CCXML or VXML scripts (and hence services) to be invoked based on OLI, TLI or physical trunk identity.
Early Media	Enables manipulation of media streams prior to answer – used for **Music Ringback**
Web-server integration	Allows CCXML and VXML to send events, timers and data to external php/cgi scripts, enabling integration with databases (for subscriber management) and billing.

Enables rapid deployment of VXML and CCXML on BICC, SIP, ISDN and ISUP

Deployed as a standalone IVR with integrated variable announcements or as part of an ecosystem with TTS/ASR servers and web Application Servers, the ARNE is available to support SS7 ISUP, ISDN, BICC and SIP, scaling from 240 to 512 RTP channels on GbE and from 240 to 1890 E1 channels on E1/STM-1.

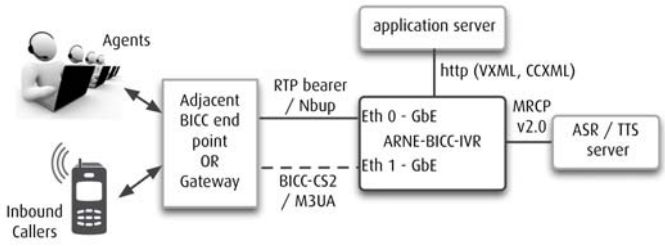
Based on FPGA technology, the ARNE IVR minimises call latencies, supporting all bearer features such as DTMF across all channels with no degradation in performance, unlike some IVR platforms that are implemented in software only.

ARNE IVR models

The ARNE IVR is housed in a compact, low maintenance robust '1u' server, which is configured at manufacture to support GbE for BICC and SIP for packet switched networks or E1/T1/STM1/OC3 for circuit switched networks. Systems can scale easily by adding 1U units as subscriber service take-up increases.

ARNE-240SIP-IVR / ARNE 240BICC-IVR

For connection to converged BICC or SIP trunked systems with up to 256 channels of G.711 RTP, including Nbup RTP encoding.



ARNE-520SIP-IVR / ARNE-520BICC-IVR

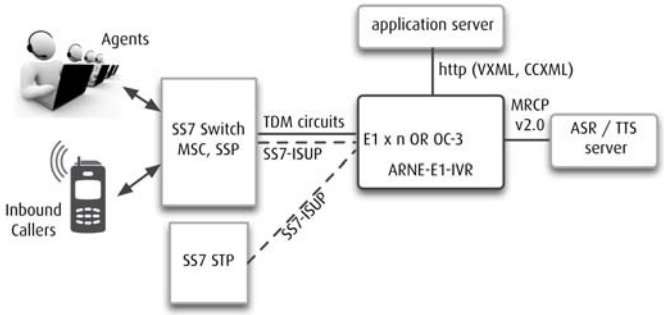
For connection to converged BICC or SIP trunked systems with up to 512 channels of G.711 RTP, including Nbup RTP encoding.

ARNE-8E1-IVR

Run-time configurable to support up to 8 E1 or T1 digital trunks for connection to circuit switched networks.

ARNE-STM1-IVR

For large multi-trunk systems, an optical network interconnect may provide a more cost effective solution. The ARNE is available with an STM-1 interconnect in a 1U chassis.



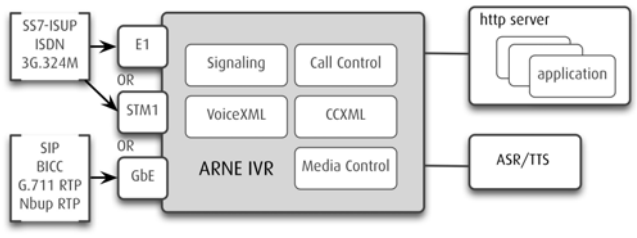
† Options are available for fault resilient builds of the ARNE including dual PSU and dual RAID HDD. Contact sales for availability and pricing.

Feature Packed Small Footprint

To deploy Self-Service applications on the ARNE IVR, all you need is to deploy the CCXML/VXML application. This can be an existing standards based application or one of the samples provided by Telesoft Technologies.

The ARNE IVR enables application developers to deploy many revenue generating Customer Care & VAS self service applications, including:

- Custom + Music Ring-back Tones
- Missed Call Alert
- Call completion services
- Pre-paid billing
- Infotainment



All triggering and service control is implemented from within CCXML and VXML, 'on-box' configuration is kept to a minimum in order to provide maximum application flexibility. Monitoring is provided through use of standard SNMP – either using a pre-installed on-box viewer or any third party SNMP client.

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. It allows voice applications to be developed and deployed in an analogous way to HTML for visual applications. Just as HTML documents are interpreted by a visual web browser, VoiceXML documents are interpreted by a voice browser, which is installed on the ARNE IVR. The ARNE IVR fetches scripts from an Application Server and executes the script on the digital voice channel.

```
<vxml version="2.0"
xmlns="http://www.w3.org/2001/vxml">
  <form>
    <block>
      <prompt>
        Hello World!
      </prompt>
    </block>
  </form>
</vxml>
```

CCXML

The ARNE implements the W3C Call Control Extensible Markup Language V1.0. CCXML providing application developers with the ability to:

- Answer inbound calls and invoke a VXML session
- Initiate outbound calls
- Connect inbound and outbound calls
- Implement 3-way conferencing
- Time events

CCXML is an event driven language designed to complement VXML dialog control and allows applications to manipulate call legs. Where VoiceXML defines user interaction (a "dialog") on a single call leg, CCXML is event driven and enables the control of call legs, initiation of new call legs and VoiceXML scripts, the joining and conferencing of call legs and the management of timed operations. The script below is a CCXML application with simple event handling:

```
<?xml version="1.0" encoding="UTF-8"?>
<ccxml version="1.0" xmlns="http://www.w3.org/2002/09/ccxml">
  <eventprocessor>
    <transition event="connection.alerting">
      <log expr="Call is Alerting"/>
```

```
    <accept/>
   </transition>
   <transition event="connection.connected">
    <log expr="Answered, now disconnect"/>
    <disconnect/>
   </transition>
   <transition event="connection.disconnected">
    <log expr="'Call disconnected'"/>
    <exit/>
   </transition>
  </eventprocessor>
</ccxml>
```

Content Play

The ARNE IVR supports concurrent play of voice announcements on every channel under the control of a VoiceXML document. Pre-recorded prompts may be stored locally, or may be retrieved from a web / content server using HTTP / HTTPS. Pre-defined on-box variable announcements allow service data variables to be voiced as digits, currencies, dates and times without the use of TTS. Announcements are stored as 8 kHz CCITT A-law, μ-law or 16-bit Linear encoded WAV files.

Text to Speech (via MRCP v2.0)

Text to Speech (TTS) engines or servers are used to convert text into audible computer-generated speech. The ARNE IVR interfaces to external TTS servers via standard MRCP V2 and RTP interfaces. This allows application developers to choose the best TTS vendor for the languages required.

The ARNE IVR supports W3C Speech Synthesis Mark-up Language (SSML) V1.0, which is used to assist the generation of synthetic speech in VXML applications. SSML provides authors of synthesizable content with a standard way to control aspects of speech such as pronunciation, volume, pitch & speed.

DTMF User Interaction

The ARNE IVR allows DTMF to be detected on every channel simultaneously, received as in band tones or RFC2833 RTP Payload for DTMF Digits.

VoiceXML allows DTMF grammars to be defined, allowing PIN numbers and credit card numbers to be collected, for example. The ARNE IVR supports 'barge-in', allowing DTMF digits to be entered before the prompting announcement has been completed and 'type-ahead', asynchronous with the prompting announcement, each digit being processed by the application in the order of entry.

Automatic Speech Recognition (via MRCP v2.0)

The ARNE IVR interfaces to external ASR servers via standard MRCP V2 and RTP interfaces. This means that application developers are free to choose the most appropriate speech recognition vendor for the languages required.

The ARNE IVR supports the XML Form of the W3C Speech Recognition Grammar Specification (SRGS) V1.0, which is used to specify grammars for use in speech recognition, allowing developers to specify the patterns of words to be listened for by a speech recognizer. The example below defines that the user is expected to voice one of three city names.

```
<grammar
  <!—Headers removed for clarity -->
  <rule id="city" scope="public">
    <one-of>
      <item>Boston</item>
      <item>Philadelphia</item>
      <item>Fargo</item>
    </one-of>
  </rule>
</grammar>
```

Record

The ARNE IVR supports the recording of media streams internally. This allows applications, such as Voice Mail and Unified Messaging, which require audio streams to be recorded for later playback, to be deployed.

Video – 3G-324M

The ARNE IVR supports Video in addition to voice sessions. Using the ARNE IVR Video capabilities application developers can deploy Video Play Back, Video Record and Interactive Caller Video Sessions.

The ARNE IVR utilises a built-in hardware-accelerated 3G-324M stack. 3G-324M is an umbrella standard for Mobile Video over telephony networks. The 3G-324M standard is the most widely used and supported standard in use today.

The ARNE IVR supports both Voice and Video calls concurrently and is capable of running up to 240 video sessions. The ARNE automatically detects incoming video calls when answering the call so the VXML application can take the appropriate action.

Video prompts can be played out to the user and incoming video can be recorded. User input can be collected in a similar fashion to voice calls.

Video files are stored in the .3gp format and are stored in the same way as .wav files. As the ARNE does not support transcoding, multiple formats of the same video file should be stored. The ARNE IVR makes the call type available within a VXML session variable:

session.connection.callType

The application inspects this variable to determine if a .3gp or .wav file should be played.

The negotiated video codec is indicated in:

session.connection.videotype

This indicates the video format to play out to the user (H263 or H264 or MPEG4)

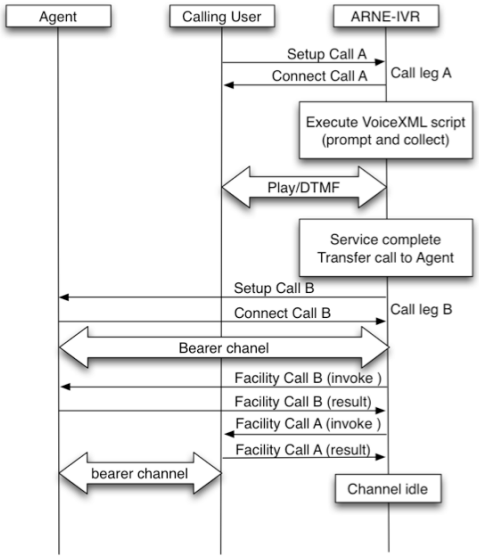
```
<!—Headers removed for clarity -->
<menu dtmf="true">
 <prompt>
  <!-- Try to play H.264 -->
  <audio src="video/h264/greet.mov">
  <!-- If H.264 fails, try H.263 -->
  <audio src="video/h263/greet.mov"
  <!-- Play audio only if all fails -->
  <audio src="audio/greet.wav" />
  </audio>
 </audio>
 </prompt>
  <choice next="Script1.vxml" />
  <choice next="Script2.vxml" />
  <choice next="record.vxml" />
  <choice dtmf="0" next="operator.vxml" />
</menu>
</vxml>
```

Agent Call Transfer

The ARNE supports call transfer such that a call can be routed to the ARNE for user interaction then forwarded on to a second destination such as a call agent. Applications include:

- Call routing applications
- Automated Call Distribution (ACD)
- Self Service transfer to an agent

The ARNE is able to use a number of different transfer procedures, including PRI-ISDN 2-B channel Transfer (2BCT) and Extended Call Transfer (ECT).



ISDN Agent Call Transfer

Call transfer can be invoked from within VoiceXML using the platform specific object com.telesoft.vxmlobjects.makecall, for example:

```
<form id="outbound_call">
  <object name="makecall"
classid="com.telesoft.vxmlobjects.makecall">
    <param name="dest" expr="session.connection.local.uri"/>
    <param name="caller" expr="session.connection.remote.uri"/>
    <param name="trunk"/>
  </object>
```

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.

Telephony attributes are configured by setting variables defined in a flat file that resides on the ARNE local storage device.

CDR's and logs

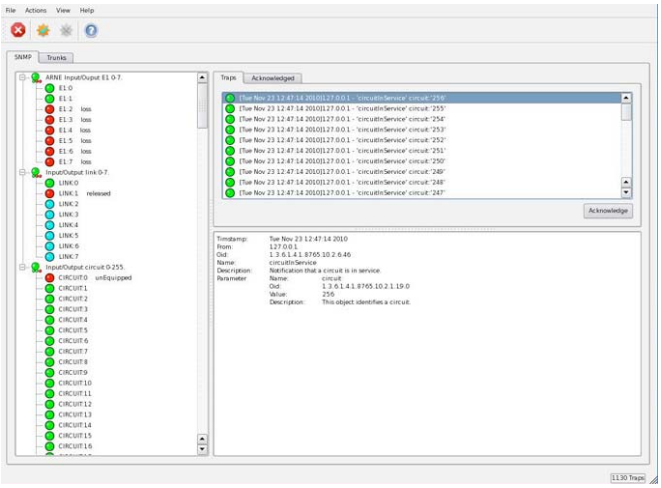
CDR's are generated according the ETSI specifications. The ARNE also handles CDR file management, however the end user should look to use FTP to transfer from the ARNE IVR older CDR's in order to prevent deletion. Full details on CDR management can be found in user guides.

Status, Alarm and Event Reporting

Status, alarm and event reporting are based on industry standard SNMP, enabling simple integration of the ARNE into any compliant third party SNMP viewer. This allows simple customisation of how status and alarms are viewed to meet the needs of each different installation.

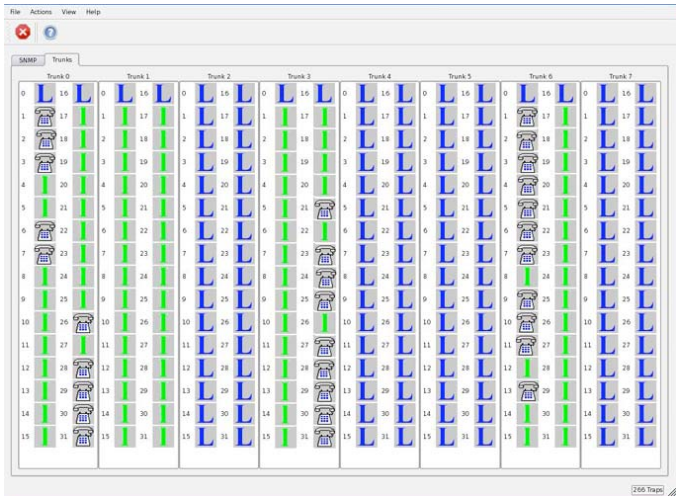
The ARNE also includes an integrated SNMP viewer that provides quick access to clear information delivered by SNMP, including physical trunk status (E1 trunks), signalling status (PRI-ISDN, SS7) and bearer channel status.

The integrated viewer provides a hierarchical view of ARNE resources, the status of each monitored resource and a historical list of SNMP traps.



Integrated Event and Status Viewer

The integrated viewer provides a view of bearer channel status, allowing an operator to quickly assess the activity on the system.

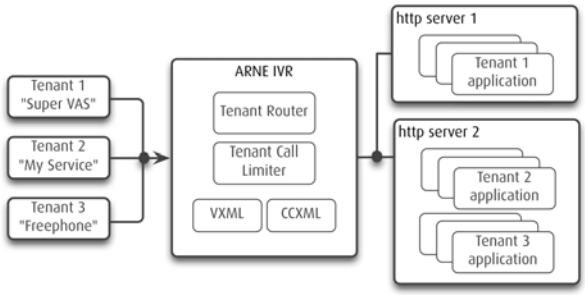


Integrated Channel Status View

Multi-tenanting

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

Each tenant service is uniquely identified by a called party number (also referred to as tli in the ARNE documentation). This number is pre-configured on the ARNE, and may be a prefix, a number range or a single unique number.



The ARNE is pre-configured with a maximum call count to limit the number of simultaneous calls that a tenant can use. If a new call is received for a particular tenant and the new call would cause the tenant's allowed call count to be exceeded, the call will immediately be rejected with reason 34 (No Circuit Available).

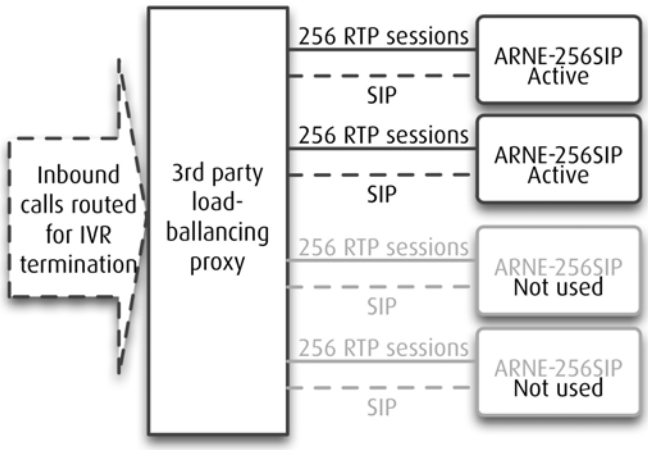
The standard ARNE configuration maps each unique tenant called party number to a specific VoiceXML or CCXML script on a specific http server. These scripts can be the same for all tenants, enabling the same service to be run by different tenant organizations, or

can be different, allowing different tenant organizations to run different services but sharing the same hardware.

Deployment Scenarios and Scaling

Due to its compact size and simplicity of operation, it is easy to add capacity to an IVR by adding ARNE units when needed.

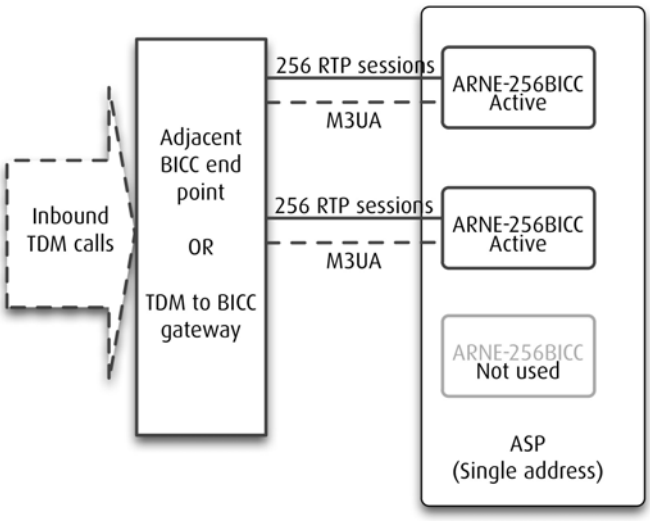
Load sharing is either automatically managed by the call control protocol or is implemented by interconnected network elements selecting bearer channels and routes appropriately.
When deploying SIP connected IVR resources, multiple ARNEs can appear to the network as a single entity by the use of a 3rd party load balancing proxy as shown in the diagram below.



Adding SIP capacity

In this configuration, additional capacity may be added by activating the unused capacity as shown below and provisioning these on the proxy.

BICC is carried over Sigtran M3UA, which provides inherent resilience, load sharing and routing capabilities. A single end point identified with a unique point code address may be split between multiple physical chassis. In such a scenario, it is the responsibility of the interconnected BICC gateway or adjacent BICC endpoint to ensure that messages associated with the same bearer channel are all sent to the same end point, and that the referenced bearer channel physically exists on that end point.

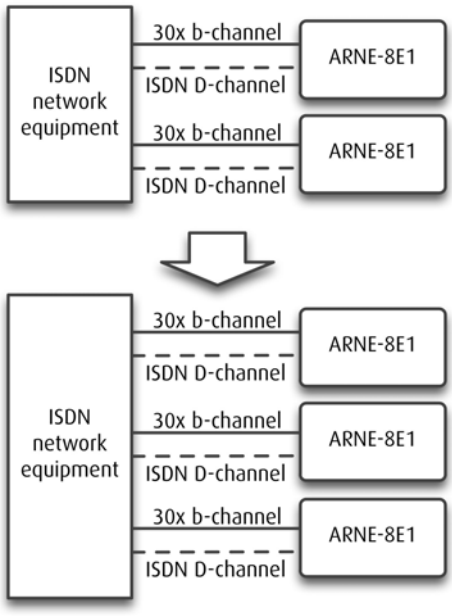


Multi-chassis BICC IVR

Systems grow by adding additional BICC ARNE IVR capacity and adding the appropriate M3UA configuration to the adjacent BICC end point or gateway.

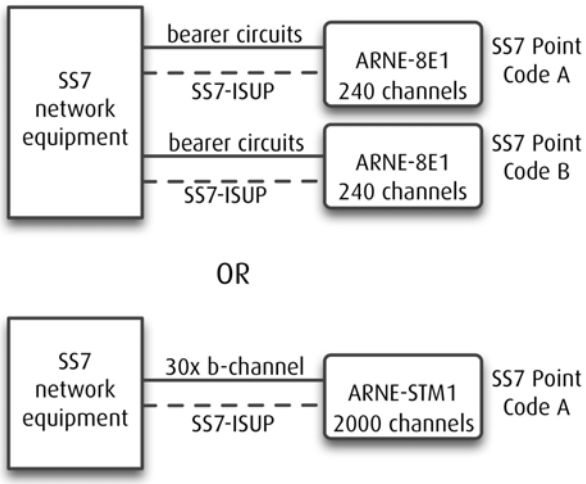
Additional resilience can be built into a BICC system using SCTP multi-homing, allowing each ARNE to provide a dual Ethernet connection for BICC traffic.

When deploying ISDN, bearer channel selection is performed by the originating network equipment that routes the calls to the ARNE. In the example below, an additional 240 channels of ISDN TDM bearer channels are added by provisioning an additional ARNE-8E1-IVR configured for ISDN.



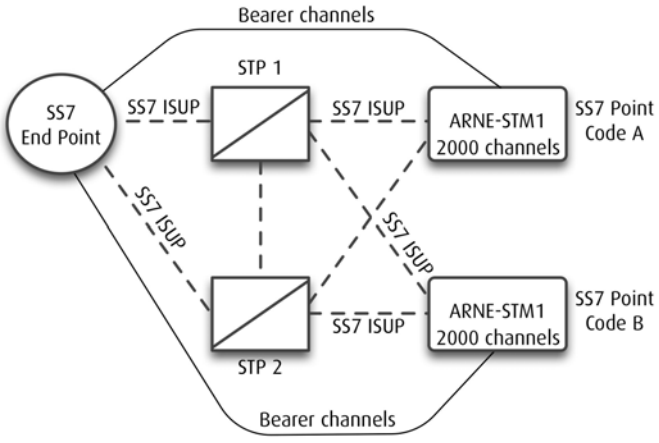
Adding ISDN Capacity in 8E1 increments

When connected with SS7, each ARNE behaves as an independent SS7 point code. SS7 is generally used for large deployments, and the number of point codes required for an IVR system may be reduced by using the higher capacity STM-1 ARNE units.



Scaling an SS7 System

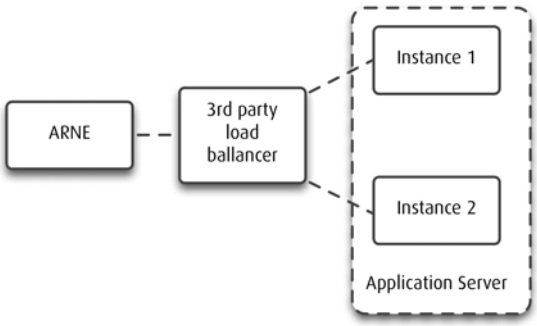
The fault resilience of such a large SS7 system can be improved by using multiple units, deployed over different sites, and interconnected with SS7 STPs, which provide multiple signalling paths.



SS7 Resilient Deployment

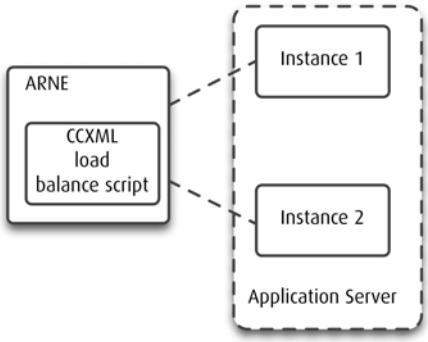
Application Server Load balancing

For mission critical systems, it may be desirable to include fault tolerance in the application server component to the IVR. This can be achieved with stateful 3rd party load balancers or DNS load balancers that share http sessions between a cluster of application servers, ensuring that all of the requests for a specific http session to the application server are routed to the same application server instance.



Application Server Load Balancing

An alternative is to implement a simple on-box CCXML script that assigns a specific application server IP address (which may be hard coded) for each new inbound IVR call.



CCXML Controlled Load Balancing

Specification – All Models (SIP, BICC, ISUP, ISDN)

Application Interfaces
VoiceXML: W3C Voice VOICEXML & CCXML
Extensible Markup Language V2.0
CCXML: W3C Call Control Extensible Markup Language V1.0
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 V2: IETF draft-ietf-speechsc-mrcpv2-12
VoiceXML / CCXML sessions on all calls simultaneously

Media Processing
Voice Play on every channel simultaneously
DTMF on every channel simultaneously
VAD on every channel simultaneously
Automated Speech Recognition via MRCP V2
Text to Speech via MRCP V2
Voice Record via MRCP V2
SSML for Speech Synthesis control
SRGS & SISR for Speech Recognition control
file://access to local remote media files
http(s)://access to remote media files
Content caching for improved latency
Built-in Grammars for Dates / Times / Currencies etc
Up to 32 languages

File formats
WAV - 16 bit a-law, 8KHZ G.711
WAV - 8 bit a-law, 8KHZ, G.711
3GPP
MPEG-4
H.263
AMR-NB

Management
SNMP

Regulatory (PSU)
USA - UL listed, FCC
Canada - CUL listed
EN 60950/IEC 60950-Compliant
Europe/CE Mark
Germany - TUV Certified

Environmental
Operating Temperature Range 10 - 35°C (50° - 95°F)
Non-Operating Temperature Range -40 - 70°C (-40° - 158°F)
Humidity Range 8 - 90% non-condensing
Non-Operating Humidity Range 5 - 95% non-condensing

ARNE-8E1-IVR

Physical
1U Rack Mount low Noise platform
PSU 200W High Efficiency
Height 1U, 1.7" (43mm)
Width 17.2" (437mm)
Depth 9.8" (249mm)
Approximate Weight 10 lbs (4.5kg)

Interfaces and Capacity
8 x T1/E1 trunks (software programmable)
ANSI T1.102, T1.403 / G.703 2,048 kbps
2 x GbE LAN
16 SS7 signalling data links
240 ISUP CIC's
240 ISDN channels

Protocols
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
EuroISDN: ETS 300-102
TR-41449, TR-41459, TR-NWT-001268, TR-NWT-001187
3G-324M (H.223, H.245, WNSRP, SRP)

ARNE-STM1-IVR

Physical
1U Rack Mount low Noise platform
PSU 200W High Efficiency
Height 1U, 1.7" (43mm)
Width 17.2" (437mm)
Depth 14.0" (356mm)
Approximate Weight 17 lbs (7.7kg)

Interfaces and Capacity
1 x STM1 optical
  Wavelength 1310nm single mode
  Connector Duplex LC style
  Reach Intermediate Reach (IR) max range 15Km
  Maximum input optical power 7dBm
  Input Sensitivity Typical -31dBm
3 x GbE LAN
16 SS7 signalling data links
2000 ISUP CIC's
2000 ISDN channels

Protocols
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
EuroISDN: ETS 300-102
TR-41449, TR-41459, TR-NWT-001268, TR-NWT-001187
3G-324M (H.223, H.245, WNSRP, SRP)

ARNE-240SIP-IVR

Physical
1U Rack Mount low Noise platform
PSU 200W High Efficiency
Height 1U, 1.7" (43mm)
Width 17.2" (437mm)
Depth 14.0" (356mm)
Approximate Weight 17 lbs (7.7kg)

Interfaces
2 x GbE LAN
240 RTP channels max (G.711)

Protocols
3GPP 24.229 IP multimedia call control protocol based on SIP and SDP Stage 3
3GPP 23.228 IP Multimedia Subsystem (IMS); Stage 2
3GPP 23.218 IM session handling; IM call model; Stage 2
IETF RFC 3261 Session Initiation Protocol
IETF RFC 3262 Reliability of Provisional Responses in SIP
IETF RFC 3263 Locating SIP Servers
IETF RFC 3264 An Offer / Answer Model with SDP
IETF RFC 2327 Session Description Protocol
IETF RFC 2833 RTP Payload for DTMF Digits...
IETF RFC 3325 Private Extensions to SIP...
IETF RFC 3323 Privacy Mechanism for SIP
IETF RFC 4048 Session Timers in SIP
IETF RFC 4240 Network Media Services with SIP (NETANN)
IETF RFC3550 RTP

ARNE-520SIP-IVR

Physical
1U Rack Mount low Noise platform
PSU 200W High Efficiency
Height 1U, 1.7" (43mm)
Width 17.2" (437mm)
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Approximate Weight 17 lbs (7.7kg)

Interfaces
2 x GbE LAN
520 RTP channels max (G.711)

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IETF RFC 3262 Reliability of Provisional Responses in SIP
IETF RFC 3263 Locating SIP Servers
IETF RFC 3264 An Offer / Answer Model with SDP
IETF RFC 2327 Session Description Protocol
IETF RFC 2833 RTP Payload for DTMF Digits...
IETF RFC 3325 Private Extensions to SIP...
IETF RFC 3323 Privacy Mechanism for SIP
IETF RFC 4048 Session Timers in SIP
IETF RFC 4240 Network Media Services with SIP (NETANN)
IETF RFC3550 RTP

ARNE-240BICC-IVR

Physical
1U Rack Mount low Noise platform
PSU 200W High Efficiency
Height 1U, 1.7" (43mm)
Width 17.2" (437mm)
Depth 14.0" (356mm)
Approximate Weight 17 lbs (7.7kg)

Interfaces
2 x GbE LAN
240 RTP channels max (G.711)

Protocols
BICC v1/Capabiity Set 1 – BICC for narrowband circuit witching
Q.1902.1 BICC PROTOCOL (CS1) FUNCTIONAL DESCRIPTION [6]
BICC v2/Capability Set 2 – BICC for IP bearer control:
Q.1902.1 BICC PROTOCOL (CS2) FUNCTIONAL DESCRIPTION [6]
Q.1902.2 BICC PROTOCOL (CS2) AND SIGNALLING SUSTEM NO 7 ISUP GENERAL FUNCTIONS OF MESSAGES AND PARAMETERS [7]
Q.1902.3 BICC PROTOCOL (CS2) AND SINGALLING SYSTEM NO 7 ISUP FORMATS AND CODES [8]
Q.1902.4 BICC BASIC CALL PROCEDURES [9]
Q.1902.5 EXCEPTIONS TO THE APM IN THE CONTEXT
IETF RFC4960 SCTP
IETF RFC 3332 SIGTRAN M3UA
IETF RFC3550 RTP
3GPP TS 29.415 NbUP

ARNE-520BICC-IVR

Physical
1U Rack Mount low Noise platform
PSU 200W High Efficiency
Height 1U, 1.7" (43mm)
Width 17.2" (437mm)
Depth 14.0" (356mm)
Approximate Weight 17 lbs (7.7kg)

Interfaces
2 x GbE LAN
520 RTP channels max (G.711)

Protocols
BICC v1/Capabiity Set 1 – BICC for narrowband circuit witching
Q.1902.1 BICC PROTOCOL (CS1) FUNCTIONAL DESCRIPTION [6]
BICC v2/Capability Set 2 – BICC for IP bearer control:
Q.1902.1 BICC PROTOCOL (CS2) FUNCTIONAL DESCRIPTION [6]
Q.1902.2 BICC PROTOCOL (CS2) AND SIGNALLING SUSTEM NO 7 ISUP GENERAL FUNCTIONS OF MESSAGES AND PARAMETERS [7]
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Please speak to a Telesoft Technologies representative if you have any requirements that cannot be fulfilled with the features mention in this document.

Where **innovative thinking** meets **engineering excellence**

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