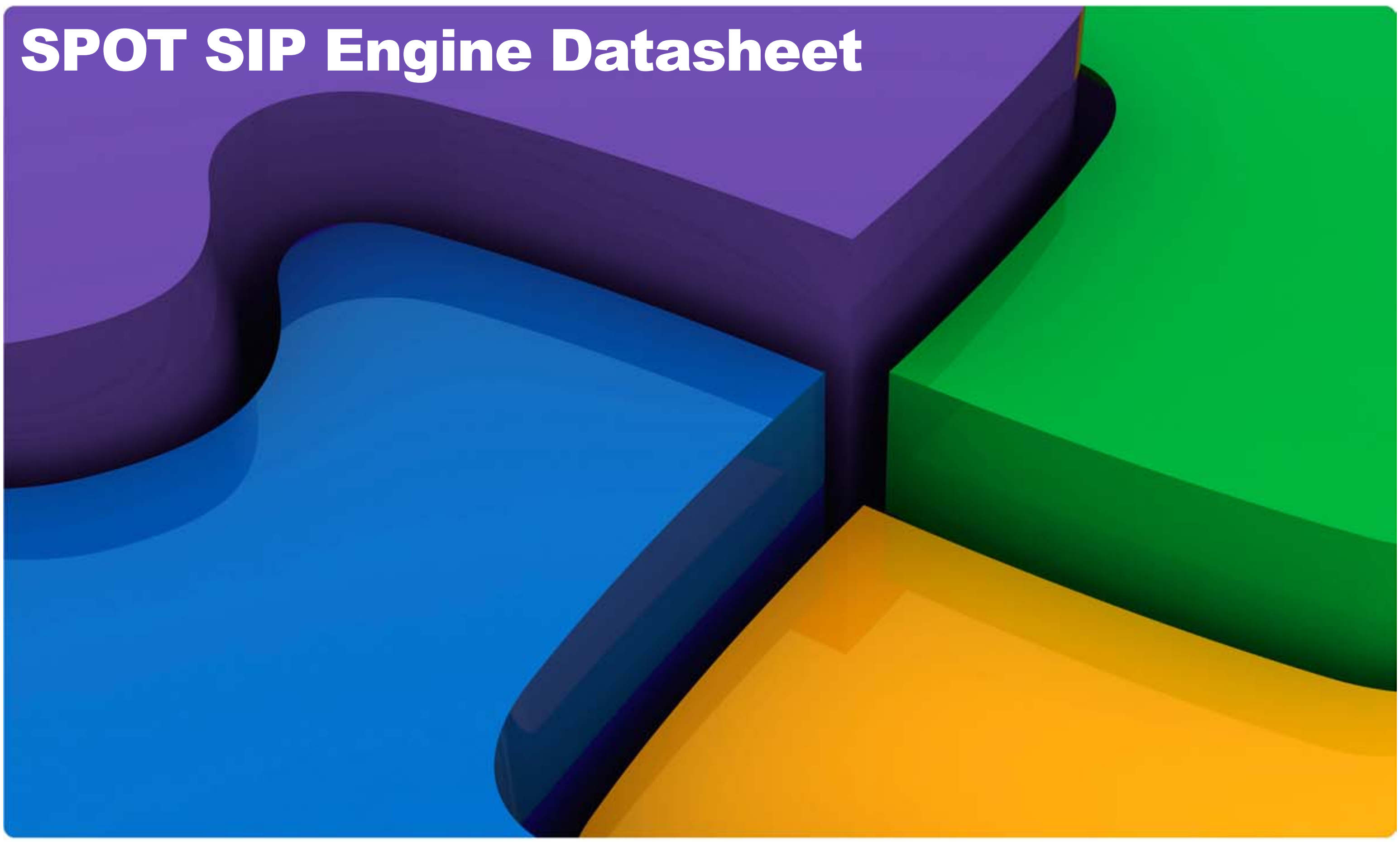
SPOT SIP Engine Datasheet



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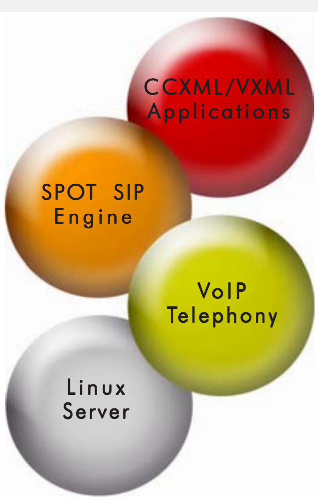
SPOT

BENEFITS

- **Software IVR/Voice Engine/IP-PBX**
  - Downloadable Linux Application
- **Drives sophisticated voice and telephony applications**
  - Both VoiceXML and CCXML Interpreters
  - Voice over IP Telephony using SIP Call control
- **Application Portability**
  - 100% "State-of-the-Standards" Compliance
- **Highest Performance, Highest Density on a single server**
  - Optimal Use of Resources/Minimal Resource Cost
  - Up to 200 CPS, even on a "whisper" application
  - Up to 1400 channel capacity (32 bit)
- **Future Proof**
  - State-of-the-art VoIP/SIP technology
- **Linear scalability**
  - Channel count as a function of memory, compute power
- **Active Redundacy**
  - Worker/Standby, or N+1 Model
  - Preserve established calls during failover
- **Vendor Independent ASR/TTS via MRCP V2**
- **T.38 fax**
- **Automatic Software Installation Checkpointing**
  - Fall-back/fall- forward
- **Browser manageable**
  - Enables cloud/hosted/remote management
  - No Linux skills required after initial server setup.
- **Flexible Licensing Options**



2009 INTERNET TELEPHONY PRODUCT OF THE YEAR

Looking for Better VoiceXML Performance?

Blazing fast and feature rich, Interact's SPOT SIP Engine enables the creation of IVR or other voice applications on a Linux server. Providing high performance, limitless scalability, and remarkable efficiencies, the SPOT SIP Engine couples both VoiceXML and CCXML Interpreters along with an integrated SIP stack and host media processing (HMP) audio capabilities to provide a "software toolbox" for creating IVR applications, IP-PBXes, MediaServers and other interactive voice systems.

Simply add the other components of your voice application - CCXML and VoiceXML script files, pre-recorded prompts (either on host or on a distinct web server), and interface whatever data base/data processing facilities that are required, and if you choose, an optional ASR/TTS engine, and your application is complete. The SPOT SIP Engine provides the rest whether the application requires inbound calling, outbound dialing, bridging, or conferencing.

Leveraging the SPOT SIP Engine's fully compliant VoiceXML and CCXML technology:

- System Integrators can provide voice applications for Fixed/Mobile carrier infrastructures, whether in IN or IMS infrastructures
- Value Added Resellers (VARs) can create voice applications/systems for their vertical markets
- Hosted IVR provider organizations can easily provide hosted voice applications for their clients such as self service applications, outbound dial applications
- Hosted IP-PBX servicers and SIP Trunk organizations can easily provide sophisticated automated attendants and/or voicemail servers, conferencing, and even the IP-PBX itself
- Enterprise organizations (IT, Call Center, other) can enable voice applications in their enterprise, whether they or another department are responsible for writing the application scripts (VoiceXML/CCXML documents)
- Web sites can deploy voice applications where voice plays a role in content delivery
- Casual users can explore VoiceXML (we recommend they use the SPOT Test Portal for that purpose)

Since Interact's SPOT SIP Engine is able to process up to 1400 voice channels on a single high performance server/host - your initial investment is reduced, and site efficiency is increased along with call processing times. Between the built in failover support which allows applications to preserve established calls in the event a server fails, and the actual performance of an application in reducing customer wait time between dialog components, the total customer experience is enhanced.

The SPOT SIP Engine enables operators to deploy cutting-edge services at a minimal TCO (Total Cost of Ownership). And given that the SPOT SIP Engine can also be used in combination with existing legacy systems, prior capital investment is protected. Simply use a Media Gateway to convert ISDN or SS7 TDM trunks to SIP Signaling and RTP Voice channels.

SPOT SIP Engine Features

- 100% "State-of-the-Standards" Compliance for Enhanced Portability
  - o VoiceXML2.0 and 2.1 Standards
  - o CCXML 1.0 Proposed Standard
  - o ECMA-262 EMAScript Language Specification
  - o ASR/TTS via MRCP V2
  - o SIP Forum's SIPconnect 1.0 (IP-PBX)
  - o IETF RFCs
- Designed for "Carrier Grade" environments and modern carrier infrastructure (IN, NGN, IMS)
  - o RHEL5/CentOS Linux servers
  - o SNMP support (optional)
  - o Redundancy/Failover support
  - o Writes system status to Syslog
  - o Performance
- Designed for high capacity production environments; can support more concurrent calls or channels/ports on a given hardware server than industry competitors
  - o Benchmarks:
    - 4 second pre-call announcement (stresses call set-up tear down),
    - Ultimate load Test (inbound/outbound dialer with\ playback, sending/detecting DTMF (RFC2833), grammar matching, and a handful of web requests.
  - o 8 core @2.3 GHz each server, 40$ idle
    - 200 CPS
    - up to 1400 channel capacity (32 bit)
    - up to 1000 channel capacity (64 bit)
- Scalable
  - o Applications scale linearly with voice channels to server capacity,
  - o Easily add additional servers as needed for the application
- Resilience/Application Redundancy
  - o Designed for error recovery and host system failover using worker/standby or the N+1 active/standby model
  - o Voice applications can be designed so there are no lost active/established calls during failover
- Configurable,
  - o Can operate in multi-tenant environments where a single server supports multiple customers and applications,
  - o Can operate in a server farm where many servers support a single integrated application
- IP Multimedia
  - o Supports a variety of audio codecs and prompt file formats along with H.263 video
  - o RTSP streams for" music on hold"
- T.38 Fax over IP
- Browser based system management
  - o Provides status and control capability, locally or remotely, of the SPOT SIP Engine and the deployed voice applications
  - o Ideal support for voice application developers - need not be Linux gurus to create/test/debug deploy voice applications
  - o Separates server management (an IT function) from voice application management (an application specialist function)
  - o Hosted providers can create/deploy a voice system, and clients can easily manage their application
- Easy voice application development
  - o Standalone test portal supporting both VoiceXML and CCXML documents
  - o Free trial license, download the SPOT SIP Engine and try on your server at your site
  - o Highly commented sample templates
  - o Both online and printable pdf documentation
- Flexible Licensing Options
- Professional Services and Support available by a highly knowledgeable and experienced entrepreneurial company

Optimized for High Density Applications

The SPOT SIP Engine was developed for use in environments that handle hundreds to thousands of voice channels. Performance testing for the SPOT SIP Engine begins at a minimum of 500 channels for a server with dual 2.4 GHz processors. This is 9X more channel support than the majority of solutions on the market and 3X+ greater than Java-based systems. SPOT features linear scalability and no limit on the number of channels that are supported in a distributed environment.

In today's evolving market, as SIP-based application servers begin to replace traditional PSTN-based services, the SPOT SIP Engine allows SIP providers to deploy next-generation service architectures. SPOT meets SIPconnect 1.0 requirements for IP-PBXs, and has the ability to send/receive faxes over IP.

Request a trial Version of the SPOT SIP Engine now and see for yourself how it compares to your current interpreter.

http://www.iivip.com/solutions/spot-sip-engine/request-a-free-trial/

SIP Call Control Features

- Inbound call acceptance
- Outbound call placement
- Call leg bridging
- Call rejection
- Call routing
- Call transfer
- REINVITE audio re-routing
- REFER support
- Registrar server support
- Proxy server support
- Authentication support
- Configurable timers
- Address restriction
- Early media support
- ENUM support
- STUN/NAT transversal support
- ISUP/SIP interworking (per SIP-I/SIP-T) for terminations/endpoints
- Access to all SIP headers (via CCXML)
- Configurable RTP port range

SIP Compatability/Interoperability

- Interoperable with the following SIP services:
    - Axvoice
    - Bandwidth.com
    - BroadVoice
    - BroadVox
    - SER/OpenSER
    - Asterisk
    - Metaswitch
- Interoperable with the following SIP Devices:
    - SJ Phone
    - Cisco/Linksys/Sipura
    - Polycom
    - Grandstream
    - X-Lite
- Interoperatable with the following Gateways:
    - Aculab ApplianX and Groomer II Media Gateways
    - TelcoBridges Media Gateway
    - Dialogics IMG1010
    - Quintum Tenor VoIP Gateways
- Sucessfully tested against SIP Forum SIPConnect V1.0 specification (PBX)

IVR Functionality

- Answers Inbound Calls
- Plays prerecorded audio prompts/announcements
- Detects DTMF tones (via RFC 2833) in RTP stream
- Records incoming audio
- Plays back recorded audio
- Places outbound calls
- Generate DTMF(RFC 2833) in RTP stream
- Bridges audio between call legs
- Transfers calls (blind, bridge, consultation)
- Streaming RTSP audio
- Streaming MP3 audio via Shoutcast and ICECAST
- Conference call management (setup, add/delete participants, announce tones, listen only or active participants, teardown)
- Conference audio mixing, record conference for later playback
- Speech Recognition (ASR)
- Speech Synthesis (TTS)
- Vendor independant ASR/TTS via MRCP V2
- Access to generic MRCP V2 recognizer headers (beyond those mapped to VoiceXML)
- Interoperable with following Speech Vendors:
    - Loquendo Speech Server (ASR/TTS)
    - LumenVox Speech Engine (ASR/TTS)
    - Nuance Recognizer/Vocalizer (ASR/TTS)
- Supports simultaneous usage of different vendor TTS and/or ASR engines
- Supports VoiceXML 2.0/2.1 IVR functionality
- Supports application server media server interface (RFC 5552)
- Supports CCXML 1.0 IVR functionality
- Scales from one to thousands of ports/channels
- Supports per channel/port, per minute and per call licensing
- Configurable RTP port range

Audio/Media Processing

- Packet loss concealment
- Fixed gain control (per channel)
- Dynamic adjustment of speed and volume on audio file playback
- Configurable jitter buffer
- Transcoding/mixing at 8, 16, or 32 KHz (as needed)
- Codec transcoding
- Voice activity detection
- Live speaker or answering machine discrimination
- Fax tones detection
- T.38 Fax send/receive
- H.263 Video Streaming to/from 3gpp files along with bridged calls

W3C/ECMA/ITU-T Standards Supported

- VoiceXML 2.0/2.1 Compliant (W3C unless stated otherwise)
- CCXML 1.0 Compliant (Implementation Report participant)
- SRGS 1.0 Speech Grammars
- SSML 1.0 Speech Markup
- SISR 1.0 Speech Semantic Interpretation
- Extensible Markup Language (XML) 1.1
- XML Document Object Model (DOM)
- XML Namespaces
- Standard ECMA-262 ECMAScript Language Specification (ECMA)
- Q.1912.5 Interworking between SIP and BICC or ISUP (ITU-T Recommendation)

Codecs Supported

- G.711 µLaw (ITU-T unless stated otherwise)
- G.711 aLaw
- G.722 ("HD Audio")
- G722.1 (Siren7, Siren14 from Polycom, wideband/ultra-wideband audio)
- G.726 ADPCM (32 bit only)
- G.729a (requires license)
- GSM (RPE-LTP Full Rate - 13kbps)
- AMR-NB (3gpp codec, requires license)
- iLBC (RFC 3952 Internet Low Bit Rate)
- H.263 Video

Audio/Video Mime Types

- audio/x-wav
- audio/x-adpcm
- audio/x-alaw-basic
- audio/x-basic
- audio/x-gsm
- audio/mp4
- audio/mpeg
- audio/x-aac
- video/3gpp

System Features

- Runs on x86 architecture
- Runs on RHEL5 CentOS5
- Delivered as a tgz file of RPMs, installs using RPM Manager
- Install utility has checkpoint/rollback capability
- "Installation verification" voice application included in initial install
- Premise or hosted solution
- SNMP support extensible to user's add-ins
- Web browser based management console
- Documented HTTP management interface
- Online documentation on SPOT SIP Engine server
- Heavily commented ExamplePack (online)
- Worker/standby and N + 1 redundancy server configurations
- Keeps active calls alive during failover
- SNMP Support (Option)
- Open platform, user can add own applications/processes
- SNMP support extensible to user's add-ins

IETF Standards

- RFC 1889 - Transport Protocol for Real-Time Applications (Obsoleted by RFC 3550)
- RFC 1890 - RTP Profile for Audio and Video Conferences with Minimal Control (Obsoleted by RFC 3551)
- RFC 2246 - The TLS Protocol Version 1.0 (Obsoleted by RFC 4346)
- RFC 2326 - Real Time Streaming Protocol (RTSP)
- RFC 2327 - SDP: Session Description Protocol (Obsoleted by RFC 4566)
- RFC 2616 - Hypertext Transfer Protocol — HTTP/1.1
- RFC 2833 - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals (Obsoleted by RFC4733, RFC 4734)
- RFC 2976 - The SIP INFO Method (Obsoleted by RFC 6086)
- RFC 3164 - The BSD Syslog Protocol (Obsoleted by RFC5424)
- RFC 3195 - Reliable Delivery for syslog
- RFC 3204 - MIME media types for ISUP and QSIG Objects
- RFC 3261 - SIP: Session Initiation Protocol
- RFC 3162 - Reliability of Provisional Responses in SIP
- RFC 3263 - SIP: Locating SIP Servers
- RFC 3264 - An Offer/Answer Model with SDP
- RFC 3265 - SIP: Specific Event Notification
- RFC 3266 - SDP: Session Description Protocol (Obsoleted by RFC 4566, Updates RFC 2327)
- RFC 3310 - HTTP Digest Authentication Using Authentication and Key Agreement (AKA)
- RFC 3311 - SIP UPDATE Method
- RFC 3323 - A Privacy Mechanism for SIP
- RFC 3324 - Short Term Requirements for Network Asserted Identity
- RFC 3325 - Private Extensions to SIP for Asserted Identity within Trusted Networks
- RFC 3326 - The Reason Header Field for SIP
- RFC 3372 - SIP-T: Context and Architectures
- RFC 3489 - STUN - Simple Traversal of UDP Through Network Address Translators (NATs)
- RFC 3515 - SIP REFER Method
- RFC 3550/STD 0064 - RTP: A Transport Protocol for Real-Time Applications (Obsoletes RFC 1889)
- RFC 2551/STD 0065 - RTP Profile for Audio and Video Conferences with Minimal Control (Obsoletes RFC1890)
- RFC 3581 - An Extension to SIP for Symmetric Response Routing
- RFC 3725/BCP 0085 - Best Current Practices for Third Party Call Control SIP
- RFC 3761 - The E.164 to Uniform resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)
- RFC 3764 - ENUM service registration for SIP Addresses-of-Record
- RFC 3842 - Message Summary and Message Waiting Indication Event Package for SIP
- RFC 3891 - SIP Replaces Header
- RFC 3892 - SIP Referred-By Mechanism
- RFC 3952 - RTP Payload Format for iLBC
- RFC 3960 - Early Media and Ringing Tone Generation in SIP
- RFC 4028 - Session Timers in SIP
- RFC 4346 - TLS Protocol Version 1.1 (Obsoletes RFC 2246, obsoleted by RFC 5246)
- RFC 4733 - RTP payload for DTMF Digits, Telephony Tones, and Telephony Signals (Obsoletes RFC 2833)
- RFC 4734 - Definition of Events for Modem, Fax, and Text Telephony Signals (Obsoletes RFC 2833, Updates RFC 4733)
- RFC 4475 - SIP Torture Test Messages
- RFC 4566 - SDP: Session Description Protocol (Obsoletes RFC 3266, RFC 2327)
- RFC 5424 - The Syslog Protocol (Obsoletes RFC 3164)
- RFC 5552 - SIP Interface to VoiceXML Media Services
- draft-ietf-speechsc-mrcpv2 - Media Resource Control Protocol Version 2

Since 1981, Interact has been dedicated to providing highly customizable Communication and Rating solutions to discriminating clients spanning the globe.

Readily equipped to supply extremely short time-to-market solutions accommodating both entry-level and central office class models, Interact builds every solution on an open, distributed, and modular architecture which is deployable on a variety of hardware platforms.

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