DiaStar® Server

The DiaStar® Server (DSS) delivers a new level of signaling, media, audio, and video transcoding services to complement and enhance the Asterisk and FreeSWITCH platforms. DSS is compliant with the Woomera open protocol and can operate in conjunction with any project or product that implements a corresponding Woomera client. The DSS allows developers access to video conferencing, video processing, call progress analysis, SS7 and SIGTRAN signaling, while continuing to develop in their native development environment.

Features	Benefits
Client/server architecture	Allows for the resources of DSS to be shared by multiple Asterisk or FreeSWITCH systems
Video Playback and Record	Allows developers to create Interactive Voice and Video Response (IVVR) systems for the delivery of complex instruction sets and other information best delivered as interactive video or graphics
Application control of video conferencing functions	Provides video conferencing with multiple screen layouts, user-types, and participant captions
Real-time audio and video transcoding	Allows real-time audio and video conferencing among endpoints that support different codecs; also enables storage of audio and video media in a single format while allowing playback on a variety of devices
SIGTRAN and SS7 signaling	Allows applications created using Asterisk or FreeSWITCH to be deployed in carrier environments where ISDN, Robbed-Bit, or other inband signaling is not available
Real-time menu builder	Enables developers to build interactive video menus from inside the application, minimizing video production requirements and providing maximum flexibility in application implementations
Abstracts and encapsulates media, signaling, and gateway functions	Allows developers to access Dialogic® functionality while working within their native programming environment

Dialogic®

Components

DSS is developed as an open source project (www.projectdiastar.org) that also includes access to closed-source components from Dialogic that provide value added functionality. These components include certain Dialogic® boards that provide ISDN variants, T1/E1 CAS, SS7, and so on, as well as certain derivatives of Dialogic's industry-leading media processing software that can add functionality, such as call progress analysis and video.

Features

- **Call Progress Analysis (CPA)** — Enabled through Dialogic® Perfect Call:
  - Provides call progress analysis that is intelligently tolerant of the wide variation in call progress signaling tones found in central offices and PBXs around the world
  - Enables accurate performance without complex programming
  - Uses DSP-based algorithms to accurately discriminate live human speech from recorded human speech and network noise
  - Allows the definition of user-defined tone detection templates
  - Makes detection results available in Asterisk DialPlan as well as the Asterisk Manager Interface (AMI)

- **SIP video**
  - Play and record audio and video
  - H.263 support
  - H.264 support
  - MPEG-4 support

- **Video transcoding between H.263, H.264 and MPEG-4**

- **Video transrating** — Allows video playback at different frame rates

- **Video scaling** — Enables video playback on screens of different sizes

- **SS7 support** — Provides conventional SS7 Support for carrier installations

- **SIGTRAN support** — Provides SS7 signaling over IP

- **G.711, G.722, G.729 codecs** — Enables flexible audio endpoint support

- **HD Voice** — Provides significantly superior natural sound and a dramatically increased sense of participation in live conversations and video conferences

- **Video conferencing** — Allows full duplex video connectivity for multi-party conferences

- **RTSP streaming** — Enables on-demand relay of RTSP video streams from RTSP servers and security cameras

- **Native media bridging** — Allows media to be bridged across the DSS (when both parties in a session are connected through the DSS), reducing load on the Asterisk or FreeSWITCH system

- **3G-324M** — Adds AMR-NB, H.263, H.264, and MPEG-4 codec support for video-enabled mobile applications

- **Fax (V.17) and FoIP (T.38) support** — Permits fax-capable applications in both TDM and IP environments (planned for a future release)

Technical Specifications

Network Interface
IP over a standard Ethernet connection
PSTN over Dialogic® HMP Interface Boards (DNI Boards)
SS7 over Dialogic® DSI SS7 Boards

PSTN Channel Density
24 spans (720 simultaneous channels) of TDM (T1/E1) network connectivity

Call Control
Protocols	SIP
	ISDN
	SS7 — ISUP and TUP
	SIGTRAN — SCTP, M2PA, M3UA, SUA, ISUP, TUP

Media Streaming
Protocols	RTCP
	RTP
	RTSP
Coders	G.711u/a
	G.722
	G.729a
	G.729b
	AMR-NB

Tone Detection Module
Interfaces Supported	SIP
	ISDN
	SS7
	SIGTRAN

Features Supported
User-defined tone detection	Up to 20 user defined tone detection rules
	Single and dual tones with or without cadence
	Tone detection from 300 Hz to 3.5 kHz
Call Progress Analysis	Busy tone detection — Detects 74 out of 76 unique busy/congestion tones used in 97 countries as specified by ITU-T Rec. E., Suppl. #2
	Ring back detection — Uses both frequency and cadence detection to detect 83 out of 87 unique ring back tones used in 96 countries as specified by ITU-T Rec. E., Suppl. #2
	Positive voice detection — Has speeds as little as 1/10th of a second
	Positive answering machine detection
	Fax/modem detection
	Intercept detection — Detects entire sequence of the North American tri-tone

Technical Specifications *(continued)*

Advance Video Module

Interfaces Supported	SIP
	3G-324M
Features supported	Play, Record
	Video transcoding
	Video conferencing
	Video overlays (Allows static images and captions to be created dynamically, simplifying the task of building video menus)
Play	Playback of voice and video, voice only
	Synchronization of voice and video
Record	Stores synchronized voice and video to a file
Video stream format	H.263 (Profile 0 level 10, 20, 30 with RFC2190 and RFC2429 packetization)
	H.264 (Baseline profile levels 1, 1b, 1.1, 1.2, 1.3)
	MPEG-4 (Simple Profile levels 0, 1, 2, 3)
Video image formats	Common Intermediate Format (CIF) PAL at 352 by 288 pixels
	Quarter Common Intermediate Format (QCIF) PAL at 176 by 144 pixels
	Sub-QCIF PAL at 128 by 96 pixels
Video frame rates	30, 15, 10, or 6 frames per second
Video file formats	Proprietary format
	Audio file (.wav/.pcm): Linear PCM 16b 8K
	Audio file (.wav/.pcm): Linear PCM 16b 16K
	Audio file (.aud): HMP native codec format (all supported codecs)
	Video file (.vid); HMP native codec format (H.263 bit-stream data; H.264 bit-stream data; MPEG-4 bit-stream data)
	Image file (.jpeg)

Video Conferencing Module

Interfaces Supported	SIP
	3G-324M

Features supported

Video and audio synchronization	1, 2, 4, 6, 9 image tiling
	Attendee captions
	Display of static image into a conference
	Display of a video clip into a conference
	Ad-hoc conferencing
	Controlled conferencing with various "roles"
	Controller or moderator
	Presenter with full duplex audio/video
	Attendee listens to and views conference, but is not seen or heard
	Number of conferees dependent on host processor

Technical Specifications *(continued)*

Supported Dialogic® HMP Interface Boards (DNI Boards)

DNI/300TEPHMP— Single span PCI with 24 T1 or 30 E1 channels

DNI/601TEPHMP —— Dual span PCI with 48 T1 or 60 E1 channels

DNI/1200TEPHMP —— Quad span PCI with 96 T1 or 120 E1 channels

DNI/310TEPEHMP —— Single span PCIe with 24 T1 or 30 E1 channels

DNI/610TEPEHMP —— Dual span PCIe with 48 T1 or 60 E1 channels

DNI/1210TEPEHMP —— Quad span PCIe with 96 T1 or 120 E1 channels

DNI/2410TEPEHMP —— Octal span PCIe with 192 T1 or 240 E1 channels

Support for Dialogic® DSI SS7 Boards

System Requirements

Hardware

Processors	Intel Xeon, Intel Dual Core Xeon, Intel Quad Core Xeon, Intel Pentium 4, Intel Pentium Extreme, and a variety of multi-core processors
Memory	1 GB recommended for voice applications
	2 GB recommended for audio/video applications
	4 GB recommended for high-density applications
Disk space	500 MB required for full installation

System

IP-only solutions	Single- or dual-processor platform with an Ethernet NIC (**Note:** 1000Base-T recommended)
Converged solutions	Single- or dual-processor PCI platform with an Ethernet NIC and Dialogic® HMP Interface Boards (DNI Boards)

Operating System

Red Hat Enterprise Linux Release 5 Update 2, 3, 4, and 5 (AS/ES/WS)

CentOS 5.5

Order Code	License	Service Contract Included	Description
G01-050-01	Base TDM	No	One channel enabled on a Dialogic® HMP Interface Board (DNI Board)
G02-050-01	Base SIP	Yes	One SIP channel
G03-050-01	Tone Detection Module	Yes	One channel of custom tone detection and call progress analysis
G04-050-01	Advanced Video Module	Yes	One channel of video
G05-050-01	Video Conferencing Module	Yes	One channel of video transcoding, transrating, and scaling
G06-050-01	Trial	No	Four channels for a 45-day trial period
G07-050-01	G.729 Codec	Yes	One channel of transcoding using the G.729a and G.729b coders
G08-050-01	3G-324M Module	Yes	One channel of 3G-324M support

Note: Included service contract is a one-year Dialogic® Pro™ Value Per Unit Plan.