



Diva softIP for SIP

Diva softIP-SIP allows any software application that is based on standard CAPI or the Diva API (Diva SDK) programming interfaces to work with IP phones and soft phones in an IP telephony environment. Development effort and time-to-market are greatly reduced so developers can focus entirely on enhancing their applications without having to make SIP protocol implementations of their own. Since Diva softIP-SIP supports key supplementary telephony services such as call hold, call transfer, and multi-party conference, as well as real-time Fax over IP according to the T.38 standard, even the most sophisticated unified messaging and call center applications can work transparently with legacy telephony as well as new IP telephony.

Diva softIP-SIP allows for a variety of configuration scenarios. As a stand-alone installation, it provides connectivity to the IP network via any standard Ethernet adapter and allows for host-based media processing. While in most cases the introduction of IP-Telephony will happen in a phased approach, softIP-SIP can also be installed in combination with any other type of Dialogic Diva telephony boards (Analog, ISDN BRI and PRI ,or E1/T1). Such hybrid configurations facilitate IP migration as application platforms can concurrently be connected to legacy systems via PBX as well as to the IP systems via IP PBX or Call Manager systems. The combination of Diva softIP-SIP and Dialogic Diva TDM boards also provides the foundation for PSTN/IP gateway applications.

Features	Benefits
Unrestricted support of Dialogic® Diva® API and CAPI	Facilitates use of Diva softIP for SIP in building IP communications solutions
Supports open standards such as SIP, T.38 FoIP, and RTP	Use of open standards is a key success factor in a move to IP telephony
Supplementary service support	Enables seamless use of common telephony services such as call diversion, call transfer, call hold, and message waiting in both hybrid and IP environments
Supports high-speed FoIP in T.38 and G.711 pass-through mode	Allows use of legacy fax machines in an IP environment with up to 33.4 kbps connection speed
Handles VoIP-specific call control	Lets developers concentrate on enhancing applications, reducing development effort and time to market
Modular design	Flexible enough for use in both hybrid environments with Dialogic® Diva® boards and in stand-alone pure IP-based media server systems
Ability to activate additional channels and features, as needed, using a flexible software licensing model	Easy scalability that allows development on a small platform followed by easy expansion to deployment to a much larger platform with minimal additional cost
Continuous compatibility testing	Ensures softIP for SIP supports industry-leading VoIP products



Technical Specifications	
API	Diva API CAPI 2.0
Functionality	Call control (SIP, SDP) Supplementary services Fax (T.30 Group 3 and T.38 FoIP) In-band DTMF and DTMF relay (RFC 2833)
Media streaming	RTP; G.711
Network interface	IP over Ethernet (NDIS-5) Dialogic® Diva® boards dependant: ISDN BRI, ISDN PRI, E1, T1, analog loopstart
Channel density	Up to 240 concurrent (TDM-IP hybrid); up to 120 concurrent (stand-alone IP) Up to 8 Ethernet or Dialogic Diva boards
Interoperability	Cisco (Unified Communications Manager, Unified Communications Manager Express, VoIP gateway) Innovaphone (gateways, IP phones) SIPPS (Soft phone) Snom (IP phone) Telebau (gateway, IP phone) Siptronic (IP phone) Kapanga (Softphone)
Application Programming Interfaces	<ul style="list-style-type: none"> • Diva API • CAPI 2.0, 4th Edition • CAPI 2.0 features <ul style="list-style-type: none"> - B-channel protocols, 64 kbps bit transparent, transparent, T.30 - Fax support T.30, MH, MR, MMR, ECM - DTMF recognition and generation (in-band and out-band) - Conferencing via line interconnect - Explicit Call Transfer with Consultation Call • IP only configuration, Host Media Processing • TDM/IP hybrid configuration
VoIP call control	Session Initiation Protocol (SIP), RFC 3261 Session Description Protocol (SDP), RFC 2327 Further SIP Methods: INFO (RFC 2976), NOTIFY (RFC 3265), REFER (RFC 3515), SUBSCRIBE (RFC 3265), REGISTER (RFC3261) with digest authentication
Supplementary services	Numbering services (abstracted as Called Party Number, Calling Party Number/CLIP, Redirecting Number) Call hold/retrieve Blind transfer Message Waiting Indication (MWI) Explicit Call Transfer (ECT)
Fax services	T.30 Fax Group 3 up to 33.6 kbps using T.38 real-time fax over IP T.30 Fax Group 3 up to 33.6 kbps using G.711 pass-through mode (requires Dialogic® Diva® Media Boards for PRI) Fax compression MH, MR, MMR Error Correction Mode (ECM)
Tone processing features	DTMF generation and recognition (In-band) DTMF relay, RFC 2833
Media streaming	IP Real-time Transport Protocol (RTP) G.711 coder, 64 kbps (64 kbps, A-law, μ -law), 10 ms, 20 ms and 30 ms packetization time
Network interface	IP over Ethernet with any NDIS-5 compatible Ethernet adapter ISDN BRI, ISDN PRI, E1, T1, analog loopstart with Diva® board
Channel density	Ethernet adapters and Diva boards can be mixed and matched Up to eight adapters (Ethernet or Diva board) per system Up to 240 concurrent channels in TDM/IP hybrid configuration Up to 120 concurrent channels in stand-alone IP configuration