SIP-Enabled Hosted PBX

HERO

OUR SOLUTION

Dialexia introduces HERO (Hosted Enterprise Remote Office), a hosted-based IP-PBX server that is a 100% SIP-based VoIP Solution. It delivers calls over IP through the customer's converged networks (voice and data). It manages SIP devices (IP hard-phones, softphones and Analog Telephone Adapters) and IP-PSTN SIP gateways. HERO is designed to work as an ASP-Model Hosted PBX for one office or in a multi-branch office through the WAN or Internet. It is placed at the Internet Telephony Service Provider's (ITSP's), Internet Service Provider's (ISP's), Competitive Local Exchange Carrier's (CLEC's) and NEXTGEN operator's premises.

Easy to manage and feature rich, HERO enables service providers to move past commoditized bandwidth and minutes of use. Service providers can reap additional revenue and become the critical communications partner of their small and medium-sized enterprise clients. Instead of investing thousands of dollars in traditional PBX phone systems, a hosted PBX allows the small/medium business to receive cost-effective phone lines with the best features around. Customers can be greeted by an automatic attendant and every phone also gets call number display, call forwarding and voice mail. With HERO you can manage all accounts from a centralized Web-based monitoring tool and know the voice communications cost for each employee.

HERO is Dialexia's solution to the search for a full-featured remote phone system with next generation technology at a truly reasonable price.

FEATURES

- **Integrated Voicemail**
- **Auto Attendant/IVR**
- **Advanced Call Forwarding**
- **Call Park/Hold**
- **Call Recording**
- **Conference Bridge**

ADVANTAGES

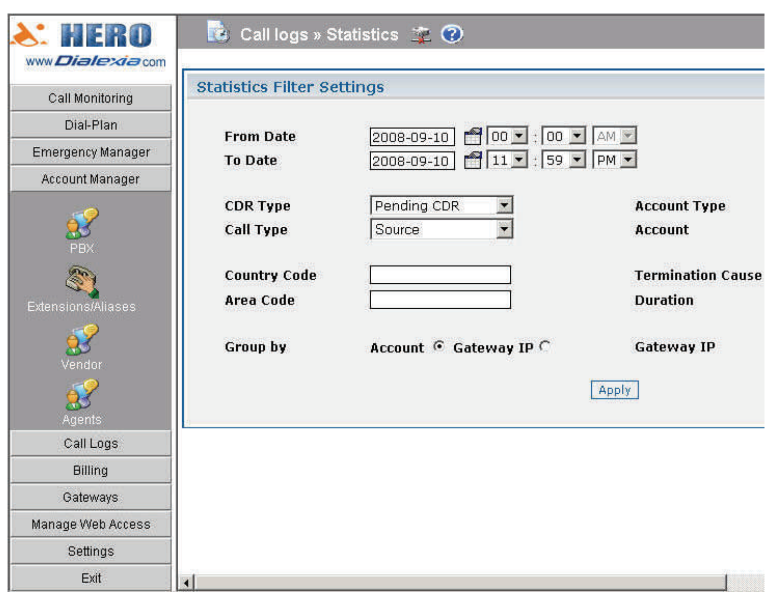
- **Reduce CAPEX by eliminating costly upgrades**
- **Reduce OPEX and increase revenue**
- **Reduce calling and moving expenses**
- **Enhance your employees' productivity**

BENEFITS

- **SIP compliant and IMS ready**
- **Lower communication costs with VoIP termination**
- **Fully interoperable with industry SIP devices**
- **Multi-branch communication & remote extensions**

ENHANCED FEATURES

- **Dial plan with Permissions**
- **Virtual Circuits Management**
- **Voicemail to Email**
- **Call Detail Reports (CDR)**
- **Unified Communications**
- **Interoperates with VoIP Service Providers for SIP trunking and termination**
- **Hunt Groups**



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HUNT GROUP

The hunt group is a time-saving call transfer distribution mechanism. User extensions are ranked under one communal extension and are assigned different call priority weights. When the communal extension is dialled, the user extension with the highest priority weight will ring. On no answer, the user extension with the second priority will ring, and so on.

ENTRY LEVEL ACD MODULES

HERO accommodates multiple ACDs and hunt groups for communal extensions. Users can be assigned to one or more communal extension groups and can be given different priority levels within each group. Users who belong to one or more groups can still generate and receive individual calls.

CALL RECORDING

Reliable and easy-to-use, conversations can be recorded (for order verification, quality monitoring, and training requirements) and saved to the user's voicemail box for future use. The result is the highest level of sound quality and the confidence that all required calls are recorded.

INTER-BRANCH OFFICE COMMUNICATION

The administrator configures a multi-branch architecture in order to bypass tolls between the branches. This is done using a multi-routing algorithm at the branch level. For example: a company XYZ has three branches based in Montreal, New York and Detroit. Each branch has its own **HERO** solution. The administrator of **HERO** in Montreal configures a route (over IP) to terminate calls for the New York and Detroit branches. With this they will not incur long distance charges. A similar routing plan can be executed at the New York and Detroit branches.

SBC/CALL RELAY (DIAL-RELAY)

Dial-Relay is an RTP processor module tool that is used by Dial-Manager for NAT-Traversal. It allows users behind Near-End or Far-End NAT to communicate. Dial-Relay has a comprehensive Web interface to monitor calls.

USER CALL PERMISSIONS

Call permission criteria, such as restricting (or enabling) long distance/international calls is created by the administrator in the call permission profile.

FAILOVER

The failover solution is used for large-scale systems (more than 300 ports). Two machines are dedicated to hosting **HERO**. The first machine is the master and the second machine is used as the backup. The master handles all call traffic to **HERO**. The backup machine remains in a standby state. Should the master server have a system failure, it is detected by the **HERO** "Watchdog" (a service interface that monitors Dial-Manager and Dial-Media for any system inconsistencies). The backup machine takes over replacing the master (to handle call traffic). The "Watchdog" sends an email alert to the administrator advising of the system failure.

PROTOCOL STANDARDS

DTMF - Out of Band (RFC 2833)
Locating SIP Servers (RFC 3263)
Offer/Answer model for SDP Codec Negotiation (RFC 3264)
Reliable Provisional Responses (RFC 3262)
RTP/RTCP (RFC 1889)
SDP (RFC 2327)
SIP Event Notification (RFC 3265)
SIP using UDP & TCP transports (RFC 3261)
Refer Method (RFC 3515)
Referred-By Header (RFC 3891)
Replaces Header (RFC 3892)
Symmetric Response Routing (RFC 3581)
Voicemail Message Indication - (RFC 3842)

DEVICE INTEROPERABILITY

Polycom (IP phones)
Cisco (IP phones, Digital Gateways)
Grandstream (FXS, FXO, IP phones)
Mediatrix (FXS, FXO, Digital Gateways)
Quintum (FXS, FXO, Digital Gateways)
Aastra (IP phones)
Multitech (FXS, FXO, Digital Gateways)
Vegastream (FXS, FXO, Digital Gateways)
Audiocodes (FXS, FXO, Digital Gateways)
Patton (FXS, FXO, Digital Gateways)

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HERO Hosted PBX

HOSTED PBX SPECIFICATIONS

Basic Features

- 100% SIP Complaint
- Web-based Java Architecture
- Voicemail
- Enhanced Billing
- Agent Option
- CDR Management Reporting
- Real-time Service Provisioning
- Invoice Creation
- AP/Analog Phone Compatibility
- Calling Plans
- Far End/Near End NAT Firewall Support
- Scalable Architecture
- Centralized Web Management
- E911 Support
- Virtual Phone Number and Virtual PBX
- DID Numbers, Provisioning and Management
- SIP Device Provisioning and Management
- Billing Plan
- Mobile Virtual Office
- Hunt Group

Advanced Features

- User Preference Setting (via Web Access)
- Auto Attendant
- Conference Bridge
- Instant Messaging (phone or PC)
- Music on Hold
- Least Cost Routing
- Internet Access and ITSP Inter-connection
- VoIP Domain to VoIP Domain Call Routing
- Tele-worker Support
- Mobile User Support
- Caller ID
- Call Transfer and Call Forward
- Call Waiting and Call Hold
- Find Me/Follow Me
- Fax to Email
- Unified Communication
- Failover Support
- Service Provider SIP Trunking Support
- PBX Administrator
- PBX End-User
- PSTN to IP Dialing
- IP to IP Dialing
- Call Relay (RTP)

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ABOUT HERO

HERO is a fully standard compliant 100% SIP-based hosted PBX software package designed primarily for ITSPs, Carriers, Cable distributors, and CLECs. However, it can also be offered to Value Added Resellers (VARs), Systems Integrators, VoIP Resellers, and VoIP Distributors who want to offer small and medium-sized enterprise (SME) customers a latest generation hosted PBX system that fully meets all of their business communications needs.

HERO is Java-based, running on Win2000/2003 Server. Its multi-platform capability is attractive to resellers that do not want to be limited to a single OS solution. It has been successfully tested with many softphones, IP phones, FXS adapters and gateways, including Quintum, Mediatrix and Cisco gateways. It can be sold bundled with any of these gateway brands. It is available in a choice of software only and/or software/hardware bundles.

Now with **HERO** SMEs will be capable of sharing the best of both worlds: unifying voice and data networks, as well as centralizing telephony networks for all their branches and remote employees.

NETWORK CONFIGURATION

HERO is a 100% SIP-based VoIP solution. It terminates calls via ITSPs or voice carriers over IP. In converged networks (voice and data), it manages SIP devices (IP hard-phones, softphones and Analog Telephone Adapters) and IP-PSTN SIP gateways. **HERO** is designed to work as an ASP-Model Hosted PBX for one office or in a multi-branch office through the WAN or Internet.

COMPONENTS & MODULES

HERO is composed of several integral components. The Dial-Manager component incorporates a SIP Proxy, a Registrar, and a Presence server. Dial-Media is the SIP IVR server. Also included are the **HERO** database and an easy to use OA&M Web interface.

Innovative add-on modules will provide you with everything you need to ensure the highest quality customer service. These include enhanced calling features like ACD and integrated voicemail, call relay (RTP) services, and failover support.

UNIFIED COMMUNICATIONS

Rules are defined for email delivery based on date/time and presence state. The unified communications system provides delivery of voicemail, and faxes to the email server.

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DIAL-MANAGER

Dial-Manager is a SIP-based proxy, JAIN-compliant software that enables real-time communication over IP networks. Dial-Manager bundles three fundamental SIP components that sit in the core of a SIP network: a SIP Proxy server, a SIP Registrar server and a Presence server.

DIAL-MEDIA

Dial-Media is an IP-based media server that runs dynamic IVR applications based on XML technology. It features:

- Seamless voicemail and media services (call park, call relay, presence, and more).
- IVR for voicemail and auto attendant.

Dial-Media's easy integration of third-party media technologies (MS Exchange, Text-To-Speech (TTS), voice recognition) provides a full range of first-class services that can be delivered to end-users.

DATABASE

The **HERO** Centralized Database collects and stores the following information:

- Dial-Manager and Dial-Media provisions
- User Settings
- SIP Session Information
- Call Detail Reports (CDR)
- Server Logs
- Error/Alarm Logs
- SIP Information Logs

OA&M WEB INTERFACE

HERO has a Web-based OA&M tool that is platform and browser independent. Depending on the user profile, it offers different access to the **HERO** database.
From an administrator level the user profiles, routing/ rate tables, and IVR server are configured. The interface also provides a dynamic platform to monitor call traffic statistics, configure gateways/domains and permissions, and to manage Web access.

The end-user level provides access to presence and call forwarding management, call log history, password changes, contact list modification, and much more.

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CONFERENCE BRIDGE

The conference bridge module enables users of the hosted PBX to invite participants (other extensions) from the same office, remote offices as well as external extensions to enter the bridge. Access to the bridge is given by an extension or a telephone number (DID or Sip trunking number). Bridge features are:

- Online scheduling/re-scheduling participants' invitations/ cancelation, with e-mail notifications
- Secure access codes to join the conference
- Real-time conference monitoring from a Web page
- Conference recording
- Customized message recorded by each participant
- Each participant can record announcements (for joining or leaving the conference) with tone alerts. The maximum number of announcements can be configured.

AUTO ATTENDANT

The **HERO** auto attendant offers an innovative IVR system with the following time-saving features: language choice, call transfer by extension or by last name, call transfer to a group (sales, support), and call transfer to the operator (by pressing "0" or staying idle). A direct external number (DID) and an internal extension number can be assigned to the auto-attendant for professional usability.

VOICEMAIL

The administrator selects how voicemail storage is divided among users by defining limits in the user profiles. This included the total number of received messages and the total disk space allowed for storage (of all messages) on the **HERO** server. Voice messages are saved as. WAV files. Users can access, save, delete, and forward voicemail through phone or standard means of file sharing such as email and network directories. Users can easily establish email notifications of voice messages (with or without the message content attached).

PRESENCE

HERO implements the SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions) standard. It is shared across applications with innovative capabilities such as status context and Instant Messaging (IM) integration. Processing logic is used to determine user settings based on their assigned permissions, preferences and presence status.
A user defines his presence status in order to broadcast his availability to others. Useful options, such as: available, not available, busy, in a meeting, out for lunch, and on vacation are a few of the presence status choices available to the user.