Q-SIPPS™ SIP Proxy Server

A call control software package, based on Session Initiation Protocol (SIP) for building scalable, reliable Voice over IP networks

Q-SIPPS provides a full array of call routing capabilities to maximize network performance in both small and large packet voice networks.

Q-SIPPS is the configuration and control element for a business telephone and multimedia communications system. It supports most of the common features found in an enterprise private branch exchange (PBX) *.

At its core, Q-SIPPS processes SIP requests and responses and provides the primary capabilities for call session management.

Q-SIPPS supports a variety of different SIP server functions, including redirect, transaction stateful or stateless proxy, and registrar.
The server is compliant with the standards described in RFC 3261.

Q-SIPPS sits in the core of a SIP network, routing calls between SIP compliant voice gateways, IP endpoints (such as IP phones), and application servers.

Q-SIPPS enables call authorization decisions to be made within the network, identifying authorized hosts before they reach terminating voice gateways.

During the setup and teardown stages of a call, Q-SIPPS can generate Call Detail Records (CDRs) in compliance with RADIUS (Remote Access Dial-In User Service) protocol and pass the information to a RADIUS server in a billing server for call billing purposes.

Full Featured SIP Proxy Server Solution

Web Server Function

A web server is included in Q-SIPPS to facilitate remote management by user from a browser via an easy-to-use user interface. To add a new user account, simply enter the user ID, name, password and account aliases. User authentication feature is also provided. Security can be enhanced by defining user classes that restrict access to gateways.

IP PBX Functions

The following features are covered under the IP PBX or IP Centrex functions. Please note that only Call Transfer, Call Forwarding, Call Hold, and Caller ID functions are included as standard features of Q-SIPPS. The other features listed are offered as optional features for extra cost.

Account Codes may be used to track and manage telecom expenses.

Anonymous Call Rejection automatically rejects incoming calls from parties who do not deliver their name or telephone number with the call.

Automatic Callback/Ring Again allows a caller who gets busy signal to dial an activation code and be automatically called back when the called station becomes idle.

Automatic Line/Direct Connect (Hotline) automatically dials a pre-assigned Centrex station's extension number or external telephone number whenever a user goes off-hook or lifts the handset.

* = optional

Call Block automatically rejects incoming calls placed from specific telephone numbers.

Call Forwarding (Busy, No Answer, Multiple Simultaneous, Variable, Selective) automatically routes incoming calls to a given extension to another preselected number under a variety of circumstances. Call Forwarding Busy forwards calls when the called extension is busy. Call Forwarding No Answer forwards calls when there is no answer after a specified number of rings. Call Forwarding Multiple Simultaneous indicates the number of forwarded calls that can occur simultaneously. Call Forwarding Variable allows users to forward all calls to their extensions to another number. There are various call forwarding options that allow differential call forwarding to be applied depending on whether the caller and/or the forwarded number are members of the Centrex group or external lines. In addition, Selective Call Forwarding allows the user to pre-select which calls will forward to a different telephone number, based on the calling party's telephone number.

Call Hold (Hard Hold) allows calls to be put on hold by dialing a feature activation code (phone does not need a Hold button). After a call is put on hold, the user may perform some task related to the call, originate another call, answer another call by using a Call Pickup feature, answer an incoming call with the Call Waiting feature, or return to a previously held call.

Call Park allows a user to place call on hold, move to a different location, and then resume the call from any other station in the Call Park group.

Call Pickup allows the lines (or a portion of the lines) in a Centrex group to be made members of a pickup group. A call ringing on any station in the pickup group can be answered from any other station in the pickup group.

Call Restrictions/Station Restrictions prevent certain types of calls from being made or received by particular stations. For example, phones in public areas can be blocked from originating calls to external numbers to prevent unauthorized users from incurring toll charges.

Call Return allows a user to originate a call to the last party or number that called the user, regardless of whether the user answered the original call or knows the caller's identity.

Call Transfer transfers an existing call to another party.

Call Waiting Originating. When a Centrex user (who is assigned the Call Waiting Originating feature) places a call to another Centrex user whose line is engaged, the calling party will hear ringing (instead of a busy signal), and the called party will hear the Call Waiting tone. If the calling user's line has Call Waiting Originating, the called user's lined does not need Call Waiting Terminating in order for that user to receive the Call Waiting tone. Upon hearing the Call Waiting tone, the called party can put the current conversation on hold to answer the incoming call.

Call Waiting Terminating alerts a user to incoming calls when the user's line is engaged on an established call. Upon hearing the Call Waiting tone, the called party can put the current conversation on hold to answer the incoming call. Different Call Waiting tones may be available to indicate whether the call is on an outside line or is part of the Centrex group. Tone Block/Cancel Call Waiting is a related feature that allows a user to disable Call Waiting tones for the duration of call so that the call is not interrupted.

Caller ID allows a user to identify the name and telephone number of a calling party before answering an incoming call. Another version of this feature --Caller ID on Call Waiting-- allows for the calling name and number to be delivered when the called party is on another call.

Calling Number Delivery Blocking prevents a caller's telephone number and/or name from being divulged to the called party (who might otherwise receive that information if they subscribe to Caller ID).

Consultation Hold calls can be put on hold by depressing the switch-hook or pressing the flash button. After completing a second call, the user is automatically reconnected to the originally held call.

Code Restriction prevents a user from dialing one or more three-digit codes. Code Restriction can be used to reduce per call charges for certain services or restrict access to long distance carriers (other than the company's pre-selected long distance carrier).

Dial Call Waiting allows a user to automatically send a Call Waiting tone to another Centrex user when the called party's line is engaged. If the calling user invokes Dial Call Waiting, the called user's line does not need Call Waiting Terminating in order for that user to receive the Call Waiting tone. Upon hearing the Call Waiting tone, the called party can put the current conversation on hold to answer the incoming call. Dial Call Waiting is activated on a per call basis, so the caller can decide to use it only when the call is important enough to interrupt an ongoing conversation.

Directed Call Park allows a user to place a call on hold, specify the extension number from which the call will be resumed, and subsequently move to that location and resume the call.

Directed Call Pickup allows a call ringing at a Centrex station to be answered at a different station. At the station where the call is to be answered, the user dials a feature code and extension number of the ringing telephone. If the user does not finish dialing prior to someone else answering the call, then the user hears a busy signal (if the Barge-In feature is not assigned) or is bridged onto to the call to form a X-way conference call (if the Barge-In feature is assigned).

Intercom Dialing allows a user to call Centrex extensions by dialing a standard 4-digit code instead of the entire 7-digit telephone number.

Hunt Groups allow calls to be redirected to other predetermined lines when the called line is busy. Hunting allows a number of lines to be grouped into a "pool" so that an incoming call is directed to whichever of the lines is available. There are a number of different hunting options, which determine how an available line is selected.

Last Number Redial allows a user to redial the last number called by dialing an access code or by pressing a single button.

Message Waiting Audible provides a user with an audible notification --a "stutter" dial tone-- when messages have been left in the company's voice mail system. Centrex service provides a Simplified Message Desk Interface (SMDI).

Message Waiting Lamp provides a user with a visual indication when messages have been left in the company's voice mail system. The indication may be a flashing lamp on a compatible telephone or on an adjacent visual message-waiting device.

Music-On-Hold provides music to callers who are waiting on hold.

Repeat Dialing automatically dials the last telephone number the user called, and, if that number is busy, continues to monitor the busy line and establishes the call when the line becomes idle.

Speed Dialing allows a user to call frequently called telephone numbers by dialing an abbreviated speed calling code instead of the entire number.

Station Message Detail Recording (SMDR) allows the corporate telecom manager to receive call detail records on a per-station basis before the monthly telephone bill is even issued. SMDR helps the customer control telephone fraud and abuse, perform accurate cost accounting, and analyze call patterns to identify opportunities for cost reductions.

Six Way Conferencing allows a user to add third parties up to a maximum of four to an existing conversation forming a six way conference call.

Toll Restriction blocks a station from placing calls to telephone numbers that would incur toll charges.

700/900 Blocking blocks a station from placing calls to 700 and 900 numbers.

Useful Functional Modules Available for Integration with Q-SIPPS

- Q-VoiceMail
- Q-AutoAttendant
- Q-Audiotext
- Q-Conference
- Q-IVR

These are all VoIP-based proven functions that are supplied with custom features to meet the customer’s specific requirement. Please discuss with our sales representative your particular requirement.

Physical Description

Q-SIPPS is built in a robust industrial grade server running Linux 9.0 operating system in a 19”-rack-mountable metal chassis. Customer may specify the capacity of the components of the server hardware to suit the intended application.

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