Perihelion Voice Server – Features

Overview

The Perihelion Voice Server consists of a dedicated appliance in the customer's premises, and runs a modified version of the highly reliable Asterisk®[1] Open Source PBX software which has seen many man-years of development. For backup and failover the Voice Server communicates with Perihelion's Central Server located in a secure Data Centre.

The Voice Server can be used with any standard IP phone handsets or soft phones, and connects to both the Public Switched Telephone Network (PSTN) and the Internet via a Voice Gateway.

Voice Server models allow for 1, 2 or 4 BRI or PRI ISDN lines, supporting up to a maximum of 120 incoming lines. Calls may also be made by using the Internet if required, providing the option of dramatic cost savings, especially for international calls. To provide flexibility, especially for expected traffic peaks (such as during marketing campaigns), it is possible to have a large number of simultaneous incoming calls over the broadband connection although the call quality may not be that of a dedicated PSTN line.

With connectivity to Perihelion's Central Server, the Voice Server provides a full set of advanced telephony functions and also offers a fully redundant backup service.

Features

The following Voice Server features are supplied as standard:

User number

In traditional office phone systems an extension number is associated with a physical object – a phone handset. With the Perihelion Voice Server, however, a *User Number* is associated with a person, regardless of where he or she may be. That person will have defined how to be contacted through a configurable *Dial Plan* which contains options such as calling an office handset, a home or other remote phone, a mobile, voicemail or any combination of these. The *Dial Plan* can be changed easily at any time via a web interface or by phone.

Dial Plans

A *Dial Plan* is a list of numbers to call sequentially. It is the heart of the system and can be used to implement a "Follow Me" system.

Whenever a *User Number* is called, either internally or from an external call, the system looks for the current *Dial Plan* for that user. Nine *Dial Plans* are available and all are customisable by the user or system administrator. Each plan can contain up to three steps, and each step can be defined as a number to call or a keyword such as Roaming, Work, Assistant, Mobile, SoftPhone, Home and Voicemail.

[1] Asterisk is a registered trademark of Digium, Inc.

By default, DialPlan1 is normally defined to call 'Work' for 10 seconds, then call 'Assistant' for 15 seconds, then call 'VoiceMail'. Thus if a call were made to User 08 the system would dial the physical phone handset for user 08 for ten seconds, then the physical handset for User 08's assistant (if there is one) for 15 seconds, and then finally the personal voicemail for user 08.

An alternative *Dial Plan* might be 'Mobile' then 'VoiceMail'. Any calls to the user's landline would be diverted to the defined mobile number. Alternatively, a user working from home could choose DialPlan3 and set this so that any incoming calls to the work number would be routed to 'Home' then 'Voicemail'.

Yet another alternative is to use a roaming SIP phone such as a WiFi-enabled mobile. This option is enabled by using the keyword 'Roaming' in the DialPlan.

For even greater flexibility, actual numbers can be used instead of keywords. The Voice Server administrator sets up the available *Dial Plans*, and they can be selected by users through the Voice Server web interface or by phone.

Groups

A Group is a list of numbers that will ring at the same time; for instance, an incoming call to an accounts team may ring all of their free handsets until one of the team answers. Up to ten Groups can be configured and a group entry can include another group.

Incoming calls

Incoming calls are handled by the system according to a set of user-configurable options.

On an incoming call the Voice Server checks the current date and time, and chooses different routes for the call depending on whether the office is open, closed or on holiday. The opening times, holidays and routing options are all user-defined.

Three stages of call routing can be defined, often in conjunction with Groups. In the default routing, for example, step one might be "call the reception phone" for 10 seconds; if there is no answer step two could be "call all the admin staff handsets" for 15 seconds and if no-one answers, go to option three which is "leave voicemail for reception".

Outgoing calls

Outgoing call options are flexible and can be changed at any time. For example, calls starting with 00 might be routed via the internet through a cheap international gateway, while normal calls starting with 0 go through the local PSTN.

Directory

The Voice Server maintains a directory of short dials which include the actual number represented by the short code and a name or description. The first 50 numbers are available to every user on the Voice Server. In the next Voice Server release the range 51-99 will be made available for personal short dials, which can also be configured on most phone handsets or softphones.

If a number is in the Voice Server Directory then any incoming call from that number will have the corresponding description displayed on the phone as the Caller ID name.

Caller ID

Any number specified in the short dial directory will be displayed on the phones as the number followed by the name specified, allowing for easy identification of incoming calls.

Messages

Messages are easily recorded and played back for use with the IVR or call queues.

Interactive Voice Response (IVR) and Auto Attendant

An IVR typically presents a caller with various choices of how the call should be routed, e.g. "Please press 1 for sales, 2 for Service or 3 for Accounts." The Perihelion Voice Server has ten IVR options available for configuration, each corresponding to a Message to be played, as well as ten options and a timeout for numbers to be called should the caller press any number from 0-9. An IVR is typically invoked on an incoming call.

Queues

A Queue represents a more sophisticated version of the Group. Ten queues can be set up and any user can be a member of one or more queues. Unlike a Group, if all the members of a particular queue are busy then the caller hears background music and after a period is told where he or she is in the queue and an estimate of how long before the call will be answered.

Hold Music

The Voice Server can play music when callers are on hold or in a queue. This is either selected at random from the pre-configured choices or the Voice Server administrator can upload any suitable mp3 file.

Holiday and work time

Working hours can be defined on a day-by-day basis and the system comes with common holidays pre-configured. These can be edited to reflect personal or company holiday variations.

Voicemail

The Voice Server includes individual user voice mailboxes which are easily accessed, and in addition voicemails can be forwarded to the appropriate user as attachments to emails, allowing them to be listened to without any delay.

Conferencing

The system supports ten 'conference rooms'. Callers dialling into one of these rooms are all able to speak to each other. The Voice Server also supports low-cost High Definition Video Conferencing as a value-added option, especially useful for companies with multiple office locations.

Call Records

The Voice Server keeps a record of all calls, logging who made the call, when, where to and how long the call lasted. The system maintains twelve months of call records. Call records are available to Voice Server administrators via the Voice Server Admin web interface for download as .xls files.

As a value-added option the Voice Server has a Call Statistics module which provides full reporting of all incoming and outgoing calls, analysed by time of day, routing, user, incoming number etc. With graphs and charts included this is particularly useful for cost and efficiency management or for monitoring Key Performance Indicators.

CRM Integration

The Voice Server integrates with a number of Customer Relationship Management systems. Incoming numbers recognised by the CRM system will trigger a screen showing details of the incoming caller (such as account name, sales activity, current products etc). Calls can be initiated from within the CRM system with a single click.

The Voice Server also allows integration with Microsoft Outlook, again allowing click-to-call from within the contacts list.

Multiple Locations

The low cost of the Perihelion Voice Server makes it easy to deploy in multiple office locations, giving customers the opportunity for even greater flexibility for backup and resilience; for instance, calls can automatically be routed through the second location if PSTN connectivity to the first location is disrupted.