



## **VoiceStream PBX**

Feature Definitions

## 2 Station / User Feature Definitions

---

### 2.1 Subscriber Station Features

| Feature Name                          | Feature Description   |
|---------------------------------------|---|
| Alternate Numbers w/ Distinctive Ring | Enables users to have up to ten phone numbers and/or extensions assigned to them. The usual ringing is provided for incoming calls to the primary phone number and users have the option of enabling a distinctive ring for calls to their second and third phone numbers. For outgoing calls from the user, the user's primary phone number is the calling line identity. This feature is often used to allow a single station to be called locally from multiple originating local calling areas.   |
| Analog Hotline                        | The Analog Hotline (sometimes referred to as Automatic Ring Down or Hot Dial) feature configures an analog telephone connected to an Analog Telephone Adaptor (ATA) to automatically call a pre-configured telephone number when taken off-hook. When configured, the telephone only makes calls to one number and cannot be used to call any other number.   |
| Anonymous Call Rejection              | Enables a user to reject calls from anonymous parties who have explicitly restricted their Caller ID. By activating the service via My Phone or dialing the Feature Access Code (FAC), callers without available caller identification are informed that the user is not accepting calls at that time. The user's phone does not ring and the user sees or hears no indication of the attempted call. This service does not apply to calls from within the group. You can deactivate the feature through My Phone or the FAC.                             |
| Barge-In Exempt                       | Enables a user to manage Barge-in settings to include allowing/not allowing users to Barge-in on your calls, and enabling/disabling the playing of a Barge-in warning tone.   |
| Business Continuity (CFNR)            | Business Continuity/Call Forwarding Not Reachable (CFNR) allows a user to designate a phone number for all calls to route to when your main phone is not reachable due issues such as unplugged phone, power outage, or natural disaster. Users have the option to activate and deactivate the service by dialing the feature code to activate or deactivate, or by configuring the service via My Phone. If activated, a user must specify the forwarding number. A status indicator on the My Phone desktop identifies whether this service is enabled. |
| Busy Lamp Monitoring                  | Provides the ability to monitor the phone status of users within a defined group (for example, busy, idle, do not disturb). Each monitored user appears on a button/line appearance, and the status of each monitored phone is indicated by the button LED lamp/status icon.  |
| Call Forward Always                   | Enables a user to redirect all incoming calls to another phone number. Users have the option to activate and deactivate the service by dialing the activate or deactivate feature access code or configuring the service via My Phone. If activated, a user must specify the forwarding number. A status indicator on the My Phone desktop identifies whether this service is enabled.<br><br>Users may also forward these calls directly to Voice Mail by using the activate or deactivate feature access code or via My Phone.                          |

|                            |  |
|----------------------------|--|
| Call Forward Busy          | <p>Enables a user to redirect calls to another destination when an incoming call encounters a busy condition. Users have the option to activate and deactivate the service by dialing the activate or deactivate feature access code or configuring the service via their web interface. If activated, a user must specify the forwarding number.</p> <p>Users may also forward these calls directly to Voice Mail by using the activate or deactivate feature access code or via My Phone.</p>  |
| Call Forward No Answer     | <p>Enables a user to redirect calls to another destination when an incoming call is not answered within a specified number of rings. Users have the option to activate and deactivate the service by dialing the activate or deactivate feature access code or configuring the service via their web interface. If activated, a user must specify the forwarding number and the number of rings before forwarding.</p> <p>Users may also forward these calls directly to Voice Mail by using the activate or deactivate feature access code or via My Phone.</p>                                     |
| Call Forwarding Selective  | <p>Enables a user to define criteria that causes certain incoming calls to be redirected to another destination. If an incoming call meets user specified criteria, the call is redirected to the user specified destination. The user controls the service via the My Phone desktop web interface in My Rules, which provides the ability to set the forwarding destination address and the criteria sets for determining which calls require forwarding. A criteria set is based on incoming calling line identity, time of day, and day of week. Multiple criteria sets can be defined.</p>       |
| Call History               | <p>Call history includes such information as start date, start time, calling number, called number, caller ID, caller ID information (city and state), call duration, and type of call (outgoing, incoming and missed).</p> <ul style="list-style-type: none"> <li>• My Account administrators can view the call history for each number assigned to their account.</li> <li>• Individual users can see their call history (outgoing, incoming and missed) and place calls via the click to dial feature.</li> </ul>   |
| Call Hold & Resume         | <p>Enables a user to place an existing call on hold for an extended period of time, and then retrieve the call to resume conversation. While the calling party is held, the user can choose to make a consultation call to another party.</p>  |
| Call Logs w/ Click to Dial | <p>Enables a user to dial a number when viewing call logs from the Call Log on a phone, My Phone web portal.</p>   |
| Call Notify                | <p>Provides an email notification to a user when they receive a phone call meeting specific defined criteria.</p>  |
| Call Redial                | <p>Enables users to redial the last number they called by clicking the feature button on their phone or by dialing the Call Redial feature access code.</p>  |
| Call Return                | <p>Allows users to call the last party that called, whether or not the call was answered by entered a feature access code.</p>   |
| Call Transfer – Attended   | <p>Attended: Enables a user to consult with the add-on party before transferring the caller. To initiate call transfer with consultation, the user hits the “transfer” button/icon and dials the add-on party. When the call is answered, the user can consult with the add-on party. To transfer, the user hangs up causing the caller to be connected to the add-on party. Alternatively, the user can use a flash hook to initiate the transfer. Users have the option to have a three way call instead by hitting the transfer button a second time after the add-on party answers the call.</p> |

|                                     |   |
|-------------------------------------|---|
| Call Transfer – Blind               | Blind: Enables a user to transfer a call to another party unannounced without conversation. To initiate a blind call transfer, the user hits the “transfer” button/icon and dials the add-on party. When the destination phone rings, the user hangs up and the call directed to the other party. Alternatively, the user can use a flash hook to initiate the transfer.  |
| Call Waiting for up to 4 Calls      | Enables a user to answer a call while already engaged in another call. When a second call is received while a user is engaged in a call, the user is informed via a call waiting tone. To answer the waiting call, the user presses the flash hook or feature button on the phone. The user connects with the waiting party and holds the original party. By pressing the flash hook/feature button, the user reconnects to the original party and holds the waiting party. The feature completes when any party hangs up.  |
| Call Waiting ID                     | Displays the Caller ID information of the waiting party in support of the Call Waiting Feature.   |
| Dialer for Chrome™                  | Users can make click-to-dial calls, access contacts, and manage selected VoiceStream PBX features directly from a Chrome browser.   |
| Directed Call Pick-up               | A Group Call Pick-Up Feature which enables a user to answer a call directed to another phone in their group by dialing the respective feature access code followed by the extension of the ringing phone.   |
| Directed Call Pick-up with Barge-in | This version of the Directed Call Pick-Up service enables the user to barge-in on the call if already answered, thereby creating a three-way call. Administrators can configure whether or not a warning tone is played when a barge-in occurs.   |
| Do Not Disturb                      | Allows users to set their station as unavailable so that incoming calls are given a busy treatment. Users have the option to activate and deactivate the service by dialing a feature access code (activate or deactivate), configuring the service via their web interface, desktop or mobile client. A status indicator on the My Phone desktop identifies whether this service is enabled.   |
| Enterprise Phone Directory          | The Enterprise Directory in My Phone – My Numbers allows Users to access Contacts and Contact Groups within their organization. The Enterprise Directory is comprised of the Users and Groups that have been provisioned in their VoiceStream PBX by their Site Administrator. Users have the ability to search Contacts / Groups by Name or Number and to initiate phone calls directly from the portal via Click-to-Call functionality.   |
| Extension Dialing, Variable Length  | Enables users to dial only the extension number to call other members of their business group, without requiring the called parties full 7 or 10 digit number to be dialed. Variable Length Extension Dialing allows a group administrator to configure extension dialing within the group to be a fixed length that is between two and six digits.   |
| Feature Access Codes                | Feature Access Codes are codes that can be entered through the phone keypad to control the configuration of certain features. To use a Feature Access Code, off-hook the phone and enter the appropriate Feature Access Code. For features that require additional input, follow the audible prompts to input the required additional information. The code list is provided in the Quick Reference Guide.  |
| Hoteling                            | Enables a user’s phone profile of phone number, features, and calling plan to be temporarily loaded onto a shared (host) phone. It is comprised of two features: Hoteling Host (Multi-user Phone) and Hoteling Guest (Transportable Profile) that work together to allow the customer to designate specific phones (hosts) that users (guests) can temporarily log into and use as their own phone. When a guest logs into a host phone, their user profile is automatically transferred to the device. The host device then becomes the user’s primary device while they are logged into it. |

|                                   |  |
|-----------------------------------|--|
| Inbound Caller ID (Name & Number) | Provides the calling name and number provided by the caller's service to the user's equipment which can be displayed if equipped.  |
| Inbound Fax to email              | Provides users the ability to receive, store, review, and manage fax messages. Users are notified of new fax messages in the same way that they are notified of new voice messages. Incoming fax messages can automatically forwarded to a user specified email address.   |
| Main Number outbound Caller ID    | All outgoing calls from user stations will reflect the Main Office Caller ID Number rather than the individual station number (Except in cases where a station has 2-way DIDs).  |
| Mobility                          | <p>A solution that allows users to unite their desktop phone with their mobile phone under one single business identity by extending the VoiceStream PBX features transparently to a user's mobile phone. Calls made from the mobile phone uses the VoiceStream PBX identity which makes it appear the user is calling from their business phone and not their mobile phone. Configuration of call settings can be made from either the mobile clients or My Phone</p> <p>The mobile phone is loaded with the VoiceStream mobile app (Android or IOS) which provides VoiceStream PBX features:</p> <ul style="list-style-type: none"> <li>• Ringing of either Desk or mobile phone or both</li> <li>• Optionally requiring answer confirmation</li> <li>• Single business appearance</li> <li>• VoIP calling over data network</li> <li>• Call-through Service using native dialer</li> <li>• Fixed Mobile Transfer - allows customers to transfer calls in progress between their landline or wireless phone and their VoiceStream PBX desktop phone</li> </ul> |
| N-Way Calling (6)                 | Allows users to add up to 5 other parties to a call. When on a call, the user presses the conference button on the phone and dials the number of the add-on party. When the called party answers, the user hits the conference button to add the add-on party. The user can repeat this sequence up to 5 times enabling a call with 6 parties.   |
| Office Anywhere                   | <p>Office Anywhere is a "Fixed Mobile Convergence" feature that provides the following functionality directly or via the Office Anywhere Portal:</p> <ul style="list-style-type: none"> <li>• Simultaneous Ring - allows customers to receive calls to their VoiceStream PBX desktop phone on any landline or wireless phones when a call is received on their VoiceStream PBX desktop phone.</li> <li>• Unified Outbound Caller ID- allows customers to place calls from any landline or wireless phone using the outbound Caller ID of their VoiceStream PBX desktop phone</li> <li>• LD and International Deflection- allows customers to place long distance and international calls from any landline or wireless phone and have the charges applied to their VoiceStream PBX bill instead of the landline or wireless phone</li> <li>• Fixed Mobile Transfer - allows customers to transfer calls in progress between their landline or wireless phone and their VoiceStream PBX desktop phone</li> </ul>  |

|                             |  |
|-----------------------------|--|
| Outbound Caller ID Blocking | <p>Enables a user to block delivery of his/her identity to the called party. The feature can be enabled for a single call or for all calls. The user controls the service via a web interface or through feature access codes. If activated, all calls made by the user have the user's identity blocked.</p> <p>If this service is activated, users can still choose to allow the delivery of their Calling Line ID on a specific call by entering the respective feature access code for Calling Line ID Delivery per Call. Once the call is over, Calling Line ID Blocking is restored</p>          |
| Personal Phone Directory    | The Personal Directory in My Phone – My Numbers allows Users to create and manage personal contacts in their web portal. Users have the ability to search personal contacts by Name or Number and to initiate phone calls directly from the portal via Click-to-Call functionality.  |
| Priority Alert              | Allows a user's phone to ring with a different cadence based on pre-defined criteria. This feature can be used as a way to quickly notify the user that a specific number is calling or when a call is from inside your group or outside your group.   |
| Privacy                     | Allows users to exclude themselves from the group and directory listings visible to other users.   |
| Push-to-Talk                | Enables user to user intercom service across an enterprise. When a user dials the respective feature access code followed by the called party's extension, the system requests that the called station answer automatically. Users and administrators can define accept and reject lists, which can include wildcards.   |
| Remote Office               | Remote Office enables telecommuters and mobile workers to use all of their features while working remotely. Since calls are still originated from VoiceStream PBX, the service provides an easy mechanism for separating personal and business phone expenses, as well as keeping alternate phone numbers private.   |
| Selective Call Acceptance   | Enables a user to define criteria that causes certain incoming calls to be allowed. If an incoming call meets user specified criteria, the call is allowed to complete to the user. All other calls are blocked and the caller is informed that the user does not wish to receive the call. The user controls the service via a web interface My Phone, which provides the ability to establish the criteria sets for determining which calls are allowed to complete. A criteria set is based on incoming calling line identity, time of day, and day of week. Multiple criteria sets can be defined. |
| Selective Call Rejection    | Enables an End User Subscriber to define criteria that cause certain incoming calls to be blocked. If an incoming call meets End User Subscriber-specified criteria, the call is blocked and the caller is informed that the End User Subscriber is not accepting calls. The End User Subscriber controls the service via a web interface, which provides the ability to establish the criteria sets for determining which calls require blocking. A criteria set is based on incoming calling line identity, time of day, and day of week. Multiple criteria sets can be defined.                     |
| Sequential Ring             | Enables users to define a "find-me" list of phone numbers that are alerted sequentially for incoming calls that match specified criteria. While the service searches for the user, the calling party is provided with a greeting followed by periodic comfort announcements. The caller can also interrupt the search to leave a message by pressing a DTMF key.   |
| Series Completion           | A Hosted POTS line feature. Every site turned up with at least two Hosted POTS lines includes Series completion at no additional charge, providing all the benefits of the traditional POTS provider hunt group, with enhanced control, and significantly less cost.   |

|                                 |   |
|---------------------------------|---|
| Shared Call Appearance          | Allows for incoming calls to ring on up to 35 additional Site phones simultaneously for a Premium Station, and 5 additional Site phones for a Standard Station, connecting the first phone to be answered. If one of the phones is already hosting an active call under the line ID, incoming calls are delivered to the active phone and any outgoing calls from another phone using the same line ID are blocked. Shared Call Appearance is used to associate a user's Mobile or Desktop client with their main number.   |
| Simultaneous Ring               | Enables users to have multiple phones ring simultaneously when any calls are received on their VoiceStream phone number. The first phone to be answered is connected. For example, calls to a user's desk phone could also ring the user's mobile phone, in case the user is not at his/her desk. An optional setting for this feature is Answer Confirmation which requires the user answering the call to hit a digit on their phone before the call is completed. This ensures that a call is not picked by a voice messaging system before the call can be answered which may happen if a call is ringing to a cell phone that is turned off. With Answer Confirmation turned and a cell phone is turned off, other phones in the simultaneous ring list will still ring and can be answered. |
| Speed Dial                      | Speed Dial functionality in the My Phone portal allows Users to program up to one hundred personal speed dial entries (00 -99) that can be dialed directly from the portal or used on their desk phones. Speed Dial entries configured in the My Phone – My Number portal are automatically accessible on the User's desk phone.  |
| Squared Key System Presentation | Squared Key System Presentation is a feature rich Key System replacement solution for the retail environment. A single Hosted Square Key station can be assigned to multiple individual lines on a single device or multiple devices simulating a traditional squared key system. Squared Key System Presentation has 4, 8, or 12 call capacity options allowing the flexibility to create small, medium, or large configurations.  |
| T.38 Fax Support                | T.38 is a protocol for relaying high-speed facsimile transmissions over IP. ATAs supporting T.38 are validated and tested; Analog faxes can be connected to an ATA line associated with a VoiceStream PBX station.  |
| Three-Way Calling               | Enables a Subscriber to make a three-way call with two parties, in which all parties can communicate with each other. To initiate a three-way call while engaged in a regular two-party call, the Subscriber presses the conference button or flash hook and dials the third party. After the third party answers, the Subscriber presses the flash hook and forms a three-way call with the two parties. To drop the third party, the Subscriber presses the flash hook and is reconnected with the original party in a regular two-party call. If the Subscriber hangs up, all parties are released.  |
| Unified Messaging               | Allows users to access their voicemails via multiple options beyond direct phone access. Voicemail notifications can be retrieved via Visual Voicemail on the MyPhone dashboard, or via email with WAV file copies sent to any email address. Additionally, if you just want a notification of a voicemail, notifications can be sent to your cell phone via text or email. This allows users to be mobile and also have real time access to their voicemails like they were in the office. Unified Messaging also supports additional the uploading of voice greetings and setting call  |
| User Web Portals                | Web tools for Site self-administration and the User Dashboard allow easy setup and manipulation of features such as Automated Attendant, Hunt Groups, Call Rules, Call History Reporting, Selective Call Forwarding, and Voice Messaging.   |
| Video (Point to Point)          | Video Calling is supported between video enabled phones and clients on the same VoiceStream system. This includes the UC-One Desktop and Mobile clients, as well as VVX 500 and VVX 600 phones using the optional VVX USB camera. The camera is plug-and-play for users with Polycom VVX 500 or 600 phones.   |

|                  |  |
|------------------|--|
| Visual Voicemail | Allows a user to retrieve and listen to their voicemails and inbound fax messages (if inbound fax configured for that user) via the MyPhone dashboard.   |
| Voice Mail       | Provides a recorded announcement message and the ability for a caller to record a voice message for the called party. The announcement message can be a personal message recorded by the voice mail owner or a default system message. |

## 2.2 UC-One App Bundle

|  |  |
|--|--|
| Microsoft Windows Support              | The UC-One desktop client is supported on Microsoft Windows desktop and laptop devices. The client is a Session Initiation Protocol (SIP) User Agent (UA) that integrates and manages various features with the <b>VoiceStream</b> PBX server platform. Using this client, the end users can easily initiate and receive phone calls from their computers or facilitate placing calls from their desk phone. |
| Apple OS X Support                     | The UC-One desktop client is supported on Apple OS X based devices. The client is a Session Initiation Protocol (SIP) User Agent (UA) that integrates and manages various features with the VoiceStream PBX server platform. Using this client, the end users can easily initiate and receive phone calls from their computers or facilitate placing calls from their desk phone.                            |
| Google Android Support                 | The UC-One Mobile app is supported for Android smart phones. Users can leverage a convenient and intuitive interface for mobile calling services. Outgoing calls placed from the Mobile Client will present the appropriate business TN Caller Id. It also allows simple management of various telephony and mobility features through the client interface.   |
| Apple iOS Support                      | The UC-One Mobile app is supported for iOS based smart phones. Users can leverage a convenient and intuitive interface for mobile calling services. Outgoing calls placed from the Mobile Client will present the appropriate business TN Caller Id. It also allows simple management of various telephony and mobility features through the client interface.   |
| Call History                           | UC One Desktop Clients and Mobile Apps save call history for placed, received, and missed calls from any device. Call history makes it easy for you to redial and call back when you miss a call or when you want to dial a contact with whom you have recently spoken.  |
| Call Settings Control                  | UC One Desktop Clients and Mobile Apps provide easy access to call settings controls for key features such as Mobility, Out of Office, and Do not Disturb.   |
| Click to Dial from Desktop Phone       | The UC One Desktop Client allows you to initiate a call with a contact as a click to dial call so that you can pick the call up with your desktop VoIP or Office Anywhere phone.   |
| Enterprise Phone Directory Integration | UC One Desktop Clients and Mobile Apps include Enterprise Directory Integration. With access to the Enterprise directory, the client enables your workforce with more efficient and timely business communications.  |
| Headset support                        | UC One Desktop Clients and Mobile Apps support a variety of wired and wireless headsets  |



|                               |   |
|-------------------------------|---|
| In-Call Controls              | <p>UC One Desktop Clients and Mobile Apps support a wide variety of in call actions to include The following call control actions are available during a voice call:</p> <ul style="list-style-type: none"> <li>• End a call</li> <li>• Mute the microphone</li> <li>• Place a call on hold</li> <li>• Open the dial pad</li> <li>• Park a Call</li> <li>• Make a new call</li> <li>• Transfer a call - Attended</li> <li>• Transfer a call – Unattended</li> </ul>                       |
| Native Mobile Network Calling | <p>When making calls on the UC One Mobile App, the end-user may choose to make a “Call-Through” outbound call utilizing the cellular data network. With the “Call Through” solution, mobile calls are made using the user's Business Identity; the call will look like it came from their main business number and not from their cell phone number. You can call enterprise contacts using their 4-7 digit enterprise extension (rather than a full 10 digit TN) using Call-Through.</p> |
| Notifications                 | <p>UC One Desktop Clients and Mobile Apps provide users with notifications to indicate Incoming &amp; missed calls, and message waiting indications.</p>  |
| Outlook Directory Integration | <p>UC One Desktop Clients can be configured through <i>preferences</i> to integrate with Outlook on your desktop computer. Searches for contacts will include contacts from your Outlook account</p>  |
| Twin to Office Phone          | <p>UC One Desktop Clients and Mobile Apps are fully “twinned” with your main business number and identity (Shared Call Appearance). All incoming calls to your business number will alert your UC One Desktop Client and Mobile App. All outgoing calls placed with your UC One Desktop Client and Mobile App will look like they were placed from your main business number.</p>   |

### 2.3 UC-One Collaboration Bundle

|                           |   |
|---------------------------|---|
| Contact Management Tools  | <p>UC One Desktop Clients and Mobile Apps allow users to easily define and organize contacts. Contacts can be individuals or user defined groups. You can share and request presence information with your contacts. You can quickly find a specific contact by using the <i>filter</i> field.</p>                          |
| Desktop Sharing (1:1)     | <p>On the UC One Desktop Client, you can share your desktop computer screen with a contact. While sharing, chat and voice calling is also available.</p>  |
| Desktop Sharing (group)   | <p>On the UC One Desktop Client, you can share your desktop with multiple contacts. You can initiate a group desktop share session, or add contacts to an existing desktop sharing session. Creating a desktop share session automatically creates a group chat session.</p>  |
| File sharing              | <p>On the UC One Desktop Client, you can offer to send a file to a contact you are in a chat session with. When offered a file through the UC One Desktop Client, you can Accept or Reject the file transfer offer. After the file transfer has completed, the received files can be opened or viewed by the recipient.</p> |
| Instant Messaging (1:1)   | <p>UC One Desktop Clients and Mobile Apps support one-to-one Instant Messaging (AKA “chat”) with your contacts.</p>   |
| Instant Messaging (group) | <p>UC One Desktop Clients and Mobile Apps support group Instant Messaging (AKA “chat”) with your contacts. You can easily add a contact to an existing chat session, or begin a chat session as a group chat.</p>   |

|                              |   |
|------------------------------|---|
| Instant Messaging History    | UC One Desktop Clients and Mobile Apps maintain local copies of your chat history.  |
| MyRoom Collaboration         | In the UC One Desktop Client, My Room is an always available and permanent room you can use to chat with anyone that joins. If you assign a conference bridge to your profile, you and your attendees can click the Call button to automatically dial in to the conference and enter the appropriate entry codes. If a conference bridge is not specified, you can initiate an N-way call in your room. The UC One Desktop Client dials out to all participants and takes everyone off hold after the last attendee answers. You can invite others to your room by dragging and dropping them from the Contacts list into the My Room Communications window. Others join your room by right clicking your name on their Contacts list and selecting Join Room |
| Outlook Calendar Integration | The UC One Desktop Client reads Outlook appointments. If there is an appointment running at the current time, and the user has enabled Outlook Calendar integration in their preferences, then their presence will be shown to their contacts as Busy – In Meeting.   |
| Rich Presence                | UC One Desktop Clients and Mobile Apps support rich presence. For each contact you have subscribed to, you can see their presence information. Similarly, your contacts can see your presence on their contact list. Icons are used to indicate your current presence status- such as online, in a call, In a meeting, busy, or idle. Your presence status is managed automatically by the system based on your activities; if desired, users may manually set their presence status.   |

### 3 Site and Optional Features

---

Site Services are non-user specific options that provide a range of capabilities generally focus on call routing options and shared use capabilities.

|                                 |  |
|---------------------------------|--|
| Account / Site Admin Web Portal | Web tools for Site self-administration and the User Dashboard allow easy setup and manipulation of features such as Automated Attendant, Hunt Groups, Call Rules, Call History Reporting, Selective Call Forwarding, and Voice Messaging.  |
| Authentication                  | Authentication is performed upon the registration of an IP phone. This ensures that the End User Subscriber of the device is authorized to gain access to the VoiceStream PBX service.   |
| Auto Attendant                  | Automated call answering and routing to the desired party or station. A recorded greeting is played to the caller after which the call is transferred to other stations or services. The Site Administrator uses a powerful web based tool to define how calls are to be transferred. Functions include: <ul style="list-style-type: none"> <li>• Allow/Disallow Direct Extension Dialing – Provides ability for callers to dial a known extension at any time to connect with the user, without requiring additional prompts</li> <li>• Enterprise Wide or Site Specific Dialing Options – Refines the list of available users and extensions to include with the Internal dialing options</li> <li>• Schedule – Defines Days/Hours for normal business operations, as well as holidays. Supporting distinct greeting messages and routing actions for Business Hours vs. After Hours.</li> </ul> |

|                                    |   |
|------------------------------------|---|
| <p>Call Center</p>                 | <p>A carrier-class, communications management product for Call Center agents and Supervisors. It is fully integrated with the VoiceStream PBX Rialto interface and hosted telephony services. Feature capabilities include:</p> <ul style="list-style-type: none"> <li>• ACD with Multiple Distribution Policies</li> <li>• Scheduling &amp; Alternate Routing Policies</li> <li>• Queuing Policies by Call &amp; Agent State (Bounced, Stranded, etc.)</li> <li>• DNIS Based Customization (up to 63 numbers)</li> <li>• Virtual Call Center             <ul style="list-style-type: none"> <li>○ Queuing</li> <li>○ Multi-location Agents</li> </ul> </li> <li>• Scheduled Reporting</li> <li>• Customizable Announcements</li> <li>• Rapid Call Center Configuration from Admin Portal</li> <li>• Agent Tools             <ul style="list-style-type: none"> <li>○ Software Client</li> <li>○ Whisper Message</li> <li>○ Disposition codes</li> <li>○ One Click Escalation</li> </ul> </li> <li>• Supervisor Tools             <ul style="list-style-type: none"> <li>○ Software Client</li> <li>○ Silent Monitoring with Barge In</li> <li>○ Real-time Agent and Queue Reporting</li> <li>○ Queued Call Management</li> </ul> </li> </ul> <p>Notes:</p> <ul style="list-style-type: none"> <li>• The main number for a call center must be a two-way number</li> <li>• Agents must have either the Premium User or Hosted PRI - User service pack</li> <li>• Supervisors must have either the Premium User or Hosted PRI – User service pack</li> </ul> |
| <p>Call Park and Call Retrieve</p> | <p>“Parks” a call using the Call Park Feature Access Code. Once a call is parked, it can be “Retrieved” at the same or another phone by using the Call Retrieve Feature Access Code.</p>  |
| <p>Call Recording</p>              | <p>Provides fully hosted solution to record, store, organize, and access recordings of customer calls. The service can be used to cost effectively address regulatory compliance obligations, monitor quality control, training, or dispute resolution. Users are assigned a call recording license, and administrators manage recordings through the My Account Portal</p>   |

|            |   |
|------------|---|
| Call Queue | <p>Designated stations logically grouped together to take incoming calls. Calls are placed into a call queue until one of the designated stations becomes available.</p> <p>A set of stations can be logically grouped together so that incoming calls ring all phones simultaneously, or ring in sequence until answered. This Site Administrator is able to edit the group in real-time to respond to changing needs or workload. Reporting is also available which enables administrators to effectively manage the queue.</p> <p>The following call queuing functionality is included:</p> <ul style="list-style-type: none"> <li>• Call queue with configurable call volume</li> <li>• Custom entrance message upload option</li> <li>• Custom comfort message upload option</li> <li>• Configurable comfort message repeat timer</li> <li>• Custom music on hold upload option</li> <li>• Reporting <ul style="list-style-type: none"> <li>○ Call Queue Stats (Overall Queue metrics) <ul style="list-style-type: none"> <li>▪ Number of Agents Staffed (avg.)</li> <li>▪ Number of Agents Talking (avg.)</li> <li>▪ Wait Time</li> <li>▪ Calls Answered</li> <li>▪ Calls Abandoned</li> <li>▪ Calls Abandoned (avg. in sec)</li> <li>▪ Calls Timed Out</li> <li>▪ Calls Transferred</li> <li>▪ Busy Overflows</li> </ul> </li> <li>○ Agent Stats (Agent specific metrics) <ul style="list-style-type: none"> <li>▪ Number of Calls Handled</li> <li>▪ Average Call Time</li> <li>▪ Number of Calls Unanswered</li> <li>▪ Total Talk Time</li> <li>▪ Total Staffed Time</li> </ul> </li> </ul> </li> </ul> <p>Call Queues must be purchased individually. Multiple individual packages can be assigned to the same site if required.</p> <p>Notes:</p> <ul style="list-style-type: none"> <li>• 2-way Number</li> <li>• Each individual hunt group uses one 2-way number and supports internal 4 digit extension dialing</li> <li>• Extension numbers 0911,1911,0311,1311 are not allowed</li> <li>• Need to support HG CLID functionality as implemented for Hunt Group</li> <li>• Call Queue Agents must be purchased and assigned to Queue</li> <li>• Call Queue Agents may be assigned to multiple call queues</li> </ul> <p>Only Site users can be assigned to Call Queues</p> |
|------------|---|

|                                   |   |
|-----------------------------------|---|
| Calling Plan Management           | <p>VoiceStream PBX provides an interface to manage the customer inbound and outbound calling plans for each site and specific user/phone numbers. Outbound Calling Plans regulate behavior for calls placed to various destination types:</p> <ul style="list-style-type: none"> <li>• Internal (intra-company)</li> <li>• Local</li> <li>• Long Distance</li> <li>• Toll Free</li> <li>• International</li> <li>• Operator Assistance</li> <li>• Chargeable Directory Assistance</li> </ul> <p>Rules for calls originating or forwarded/transferred from a site or number are as follows:</p> <ul style="list-style-type: none"> <li>• Allow. Calls may be placed to destination type.</li> <li>• Allow with Authorization code. User must enter a predefined code to place the call to destination type.</li> <li>• Block. Calls are blocked to the destination type.</li> <li>• Transfer to a Number. Calls attempted to the destination type are transferred to a specified number.</li> </ul> <p>Inbound calling plans control call flows originating from internal sources as well as calls forwarded/transferred from external sources. Rule options for inbound calls are as follows:</p> <ul style="list-style-type: none"> <li>• Allow or Block inbound calls originating internally.</li> <li>• Allow or Block all inbound calls from external sources.</li> <li>• Allow transferred external calls only.</li> </ul> |
| External Calling Line ID Delivery | This feature allows subscribers to be able to view the caller ID information of users from an external site/enterprise.   |
| Group Call Park / Pickup          | Allows for the management of Call Park/Pickup groups. Members can park calls against other available members of the Call Park group by dialing the feature access code. When initiated, the call is parked against the first available member of the group. The parked call remains on hold until it is picked up (recalled) by a member of the group. Members of a Group Call Pickup can pick up (answer) incoming calls to any other member of the same Call Pickup group.  |
| Hunt Group                        | A Hunt Group is a set of stations can be logically grouped together so that incoming calls ring all phones simultaneously, or ring in sequence until answered. The Site Administrator is able to edit the group in real-time to respond to changing needs or workload. The number of hunt group needed is specified at the time of order.   |
| Group Paging                      | A site level feature which allows a user to initiate a unidirectional page to a predefined set of users by dialing an assigned phone number or extension. The set of users allowed to initiate a page must also be predefined. The maximum number of targets is 75 users. Users in a Page Group can be assigned across an Enterprise.   |
| Intercept (Group and User)        | The Intercept feature set provides VoiceStream PBX Service Provider (SP) administrators with call intercept capability using call routing and announcement options. This service can be leveraged for service suspension needs, as well as in support of CALEA requirements. The service can be assigned to an individual user's phone number (for example, when they have left the company) or it can be assigned to all the members of the group. End-customers and their users will not see these features.  |
| Internal Calling Line ID Delivery | This feature allows subscribers to be able to view the caller ID information of another user within the same site/enterprise.   |

|                        |  |
|------------------------|--|
| Meet Me Conferencing   | <p>An integrated feature allowing multiple callers to join in a single call which enables meetings among diversely located invitees at any time. Features include:</p> <ul style="list-style-type: none"> <li>• Support for both direct dial-in and toll free numbers</li> <li>• Multiple conference bridges per Site</li> <li>• Only Site Users can be assigned as moderator users</li> <li>• System generated moderator PINs and conference ID (participant PINs) delivered via email</li> <li>• Site administration Controls through My Site</li> <li>• Moderator Control via My Phone dashboard</li> <li>• Moderator In Call Controls via DTMF</li> </ul>  |
| Music on Hold          | <p>Pre-recorded music or announcements are played to callers when placed on hold. A default music program is provided for a site. If custom programming is desired for a site, an audio file in .wav format meeting specifications can be uploaded for each site manually.</p>   |
| Office Anywhere Portal | <p>The Office Anywhere portal is a virtual user/service that handles incoming calls from Office Anywhere locations and prompts for the destination address. The Office Anywhere portal works in conjunction with the user service Office Anywhere. The portal serves a purpose similar to a Two-Stage Dialing service. Once the destination digits are collected, the call proceeds as if the call originated from the user's desk phone.</p>  |
| Receptionist Client    | <p>The Receptionist is a thin client Telephony Attendant Console specifically developed for hosted environments. It is used by "front-of-house" receptionists or telephone attendants, who screen inbound calls for enterprises. Receptionist provides the following functionality:</p> <ul style="list-style-type: none"> <li>• Web-based interface, accessible from a web browser.</li> <li>• Ergonomic design that follows the natural work "flow" of a call from the left to the right of the screen</li> <li>• Professional call handling; critical information is available "real time"</li> <li>• Accurate delivery of messages through a one-step process when people are unavailable</li> </ul> |
| VLAN Tagging Support   | <p>VLANs provide a mean to separate traffic within the customer LAN, allowing them to apply distinct logical networking policies. VLAN's must be configured on the customers LAN in advance of turning up any phones, and the tag for the phones must be defined in advance.</p>   |
| Voice Portal           | <p>The voice portal provides an interactive voice response (IVR) application that can be called by members of the group from any phone, to manage their services and voice mailbox, or to change their passcode.</p>   |