KB - 2025 - PBX Set up Guide

1. Phase 1: Carrier Services: Setup Client, Porting DIDs, & Registering DIDs

Complete See
Number Porting: [redacted]

Register with Free Caller Registry: [redacted]

Client Endpoint Setup: [redacted]

2. Phase 2: Set up the PBX

1. Update VoIP Spreadsheet
    1. Pick the IP address we will use for the PBX VM install
    2. Determine the Registration Code we will use for the PBX if remaining
    3. Determine the Public IP of the Physical Office where the phones will end up
2. Set admin future password
    1. 12+ characters, random generated, document in ITGlue under Grexo VOIP > Passwords
3. While PBX is being created
    1. Set up Host name
        1. Log into [redacted]
    2. **Account > My Products > All Products > Manage Products >** [redacted] **> DNS > Manage DNS**
        1. Add new Records
            1. Type: a
            2. Name: {3/4 letter host}
            3. Value: IP address of PBX determine in Phase 1
            4. TTL: ½ hour
            5. Save
        2. Confirm set up of [client].[redacted].com
4. Once PBX is Created
    1. Log in as Admin
    2. Time Zone
        1. Server > System Clock
        2. Verify time is correct (go manual if needed)
    3. Network settings
        1. Server > IP Configuration > Allow Nat Port Forwarding "Yes"
            1. External IP Address (You can click "Look up External IP")
            2. Advanced Options > Hostname = [client].[redacted].com (step 3a)
    4. Apply registration code
        1. Server > Registration
            1. PBX Information Registration Code
        2. Under Business Information
            1. Use client's business name and address
        3. Under Contact Information.
            1. Full name: [redacted]
            2. Email Address: [redacted]
            3. Phone: [redacted]
        4. Apply PBX Information/Registration Code
5. Access Control Rules
    1. Server > Networking > Access Control
        1. IP Blocking Options (set all to 5)
            1. Set all Values to 5
            2. Yes Share Blocked Ips with Digium
            3. Save
        2. All Networks
            1. All values set to "Off"
    2. All Office Locations should be ***"Office Description ISP"***
       All remote users should be ***"Remote User or Location"***
       Examples:
       Office [redacted]
       Office [redacted]
       Office [redacted]
       Remote [redacted]
       Remote [redacted]
        1. Office [redacted]
            1. Network is Public IP
            2. All Options = Yes
        2. Office Client
            1. Network is Public IP
            2. Never Block IPs = Yes
            3. Web Admin = ***No****
            4. Admin API = No
            5. Printing, SIP, Web User Portal, User API, NTP, SNMP = Yes
        3. Remote UserName
            1. Network is Public IP

        2. Never Block IPs, Web Admin, Admin API, Printing, User API, NPT, SNMP = NO
        3. SIP, Web User Portal = Yes
    3. ***If the client has their own admin account for running reports, this option can be yes with a documented ticket number in the rule name.***
6. Apply *.[redacted].com SSL certificate
    1. PEM Values are found here: https://[redacted]
    2. **Server > HTTPS and Proxy** Turn off "Switchvox Managed?"
    3. Copy and paste text from each format into each field exactly. Make sure there are no trailing spaces or returns. Text must be exact. Paste as Plain Text.
        1. X.509 Certificate in PEM Format
        2. RSA Private Key in PEM Format
        3. Intermediate CA Certificate in PEM Format
    4. Server > Phone Networks
        1. Edit All Neworks
            1. Primary Host PBX
                1. Change Host address to [client].[redacted].com
            2. Advanced > Enable UDP Persist-Connection set to "yes"
            3. UDP Persist-Connection Internal set to "30"
        2. Edit Internal
            1. Primary Host PBX
                1. Change Host address to [client].[redacted].com
            2. Advanced > Enable UDP Persist-Connection set to "yes"
            3. UDP Persist-Connection Internal set to "30"
7. Log into the system by checking [client].[redacted].com
    1. Create additional admins:
        1. **PBXAdmin**
            1. Use current password in ITGlue
            2. Uncheck "Force change password on next login"
            3. Give all permissions
        2. **Pbxsupport**
            1. Use current password in ITG,
            2. uncheck "Force change password on next login"
            3. Set permissions based on ITG document "PBXSupport Account Permissions"
                1. Call Routing: Read Only
                2. Admins: No Permission
                3. Networking:
                    1. IP Configuration – Read Only
                    2. HTTPs and Proxy – Read Only
                    3. All others – Read/WRite
                4. Save for the above. All other permissions can be set to full "Read/Write"
        3. **Pbxmanage**
            1. Use current password in ITG
            2. Uncheck "Force change password on next login"
            3. Only Extensions, Diagnostics, and Maintenance should be set to "Read Only" All others should be set to "None"
            4. ***IMPORTANT* You must log in as PBXManage and accept the license**
    2. Look for an apply any remaining updates
        1. Log in as **PBXAdmin**
        2. **Server > Maintenance > Updates**
8. Setup FTP Backups
    1. Server > Backups
    2. Set up Automatic Backups
    3. Automatic backups will occur every: Day @ {pick an after hours time}
        1. Number of Backups to keep: 3
    4. See IT Glue for FTP Host, User, Password, Path
    5. FTP Directory Path /VoIPBackup/[IP Address]
        1. Add Permissions for "switchvox" user
        2. Give "allow" for all but "Full Control"
    6. Make sure to add the path to the NAS
    7. Set the following Directories Available for Backup to "YES":
        1. Sangoma Phone Idle Screens
        2. Voicemail and Fax Files
        3. Voicemail Greetings
        4. Music On Hold Files
    8. Test FTP Settings
    9. Run First Backup
        1. You can leave everything default at this time. There shouldn't be voice mail, greetings, hold music, etc, just yet.
    10. Check the Nas /VoIPBackup/[IP Address] folder that the back up completed
9. SMTP Settings
    1. Tools > Notifications > SMTP Settings (IT Glue)
        1. Email "From" Address: [redacted]
        2. Outbound SMTP Server[redacted]
        3. Outbound SMTP Username: (blank)
        4. Outbound SMTP Password: (blank)
        5. Outbound SMTP Port: 25
        6. Connection Encryption: STARTTLS
        7. **Note: This sends email through Proofpoint. It will not relay mail unless the "From" address is set specifically to** [redacted]
    2. Save
    3. Send Test Email to: [you]@[redacted].com
       Click Run Diagnostic Tool
    4. Confirm Email
10. Configure extension template and import spreadsheet with all users and extensions.
    1. Create CSV
        1. Blank CSV file found in: [redacted]
        2. Ext, fname, lname, login_email, email (for voice mail)
        3. Set Password to: [Client]#[Compnay Zip]
        4. Set voicemail pin to: 0+[redacted]
        5. Leave the rest blank, but they must be included
    2. Import CSV
    3. Turn off Password
    4. Override Dial Plan: Yes. Dial plan[redacted]
11. Set Temporary DID
    1. SIPStation setup

1. VoIP Setup > VoIP Provder
    1. **<u>Sip Provider Information</u>**
        1. **SIP Provider Name: VI Organization**
            1. Your Account ID: [redacted]
            2. Your Password: Make one up and document it
            3. Host/IP Adress: [redacted]
            4. Callback Ext: Should be a default Ext on your system
            5. DTMF Mode: [redacted]
        2. Peer Settings
            1. Host Type: Peer
            2. Apply Incoming Call Rules to Provider: Yes
        3. Caller ID Settings
            1. Supports Changing Caller ID: yes
            2. Caller ID Name: Blank
            3. Caller ID Number: Main Office Number
        4. Connection Settings
        5. IP PORT: [redacted]
            1. SIP Expiry: 120
            2. Proxy Host: Blank
            3. Authentication User: [redacted]
            4. Always trust this provider: Yes
            5. Qualify Host: Yes
            6. SIP Transport: UDP
            7. SIP Provider Host List
                1. Enter IP and click "+"
                2. [redacted]
                3.
                4.
                5.
                6.
        6. SIP Provider Name: VI Termination
            1. Your Account ID: Grexo
            2. Your Password: Make one up and document it
            3. Host/IP Adress: [redacted]
            4. Callback Ext: Should be a default Ext on your system
            5. DTMF Mode: [redacted]
        7. Peer Settings
            1. Host Type: Peer
            2. Apply Incoming Call Rules to Provider: Yes
        8. Caller ID Settings
            1. Supports Changing Caller ID: yes
            2. Caller ID Name: Blank
            3. Caller ID Number: Main Office Number
        9. Connection Settings
            1. Qualify Host: Yes
            2. SIP Provider Host List
                1. Enter IP and click "+"
                2. [redacted]
                3.
                4.
                5.
                6.
                7.
                8.
                9.
                10.
2. Set up > Call Routing > Incoming Calls
3. Create Single DID Route
4. DID Management – DID Results

12. Outgoing Calls > Outgoing Call Rules
    1. 911 Number exactly matches 911 Sip Provider/Sip Station1
    2. Internal Any
    3. Local 10
        1. Pattern to Match
            1. Number Begins with… [blank]
            2. The rest of the numbers must be between 10 and 10 digits in length
            3. Before connecting the call, trim 0 digits from the front, and then prepend the digits 1
        2. Call through
            1. Primary… SIPStation 1
            2. Add Failover Call Through Provider >
                1. Failover… SIPStation 2 – SIPStation 6
            3. Is this rule final? YES
    4. Local 1 + 10
        1. Number begins with the digits 1
        2. The rest of the number must be between 10 and 10 digits in length.
        3. Before connecting the call, trim 0 digits from the front, and then prepend the digits [blank] to the number.
        4. Call through
            1. Primary… SIPStation 1
            2. Add Failover Call Through Provider >
                1. Failover… SIPStation 2 – SIPStation 6
            3. Is this rule final? YES
    5. Toll Free
        1. Number begins with the digits 1(800|888|877|866|855|844)
        2. The rest of the number must be between 7 and 7 digits in length.
        3. Before connecting the call, trim 0 digits from the front, and then prepend the digits [blank] to the number.
        4. Call through
            1. Primary… SIPStation 1
            2. Add Failover Call Through Provider >
                1. Failover… SIPStation 2 – SIPStation 6
        5. Is this rule final? YES

6. Long Distance
    1. Number begins with the digits 91
    2. The rest of the number must be between 10 and 10 digits in length.
    3. Before connecting the call, trim 1 digits from the front, and then prepend the digits [blank] to the number.
    4. Call through
        1. Primary… SIPStation 1
        2. Add Failover Call Through Provider >
            1. Failover… SIPStation 2 – SIPStation 6
    5. Is this rule final? YES
7. 988
    1. Number begins with the digits 988
    2. The rest of the number must be between 0 and 0 digits in length.
    3. Before connecting the call, trim 0 digits from the front, and then prepend the digits [blank] to the number.
    4. Call through
        1. Primary… SIPStation 1
        2. Add Failover Call Through Provider >
            1. Failover… SIPStation 2 – SIPStation 6
    5. Is this rule final? YES
8. Fix 10-digit dialing
    1. Note: stripping the 1
    2. Number begins with the digits 1
    3. The rest of the number must be between 10 and 10 digits in length.
    4. Before connecting the call, trim 1 digits from the front, and then prepend the digits [blank] to the number
    5. Call through
        1. Primary… SIPStation 1
        2. Add Failover Call Through Provider >
            1. Failover… SIPStation 2 – SIPStation 6
    6. Is this rule final? YES