FreeStyl SIP Quick Reference Sheet

A number of basic set-up documents and videos are on the product webpage for this model. Please refer to these resources or your professional installer to avoid having to wait for a customer service representative.

Basic set-up instructions are within the FreeStyl SIP User's Manual available in the Resource section of the FreeStyl SIP product pages.

BASIC TROUBLESHOOTING

If you encounter any issues with the system, first POWER CYCLE the base station and if needed the handset(s).

If the problem continues, try resetting the base station and/or depending on the issue, the handsets as well.

NOTE: Resetting a handset will cause the handset to lose registration and all stored information and resetting the

base station will cause loss of all settings and stored configuration.

* No connection to SIP server - make sure DNS is populated (can use 8.8.8.8 / 8.8.4.4).

* "Dynamic Port Enable" – In the base station GUI, enabling this feature may solve issues where the network requires the transport type to change from UDP to TCP. Issues such as one-way audio or dialing outside using longer number of digits doesn't work but short numbers do, can be solved by turning on this feature.

*Download latest base station firmware: https://www.engeniustech.com/engenius-products/multi-line-phone-system-freestyl-sip/

TIPS

Default IP address of the base station: 192.168.1.156 with both user name and password = admin

Customize the display on the handset: It is recommended that you customize the name of each handset to match the user's name and/or extension the handset is using. Press menu, 5 (phone settings), then 7 (custom name), then enter the desired information.

How to transfer a call to a different PBX extension: For most PBX systems, while on the call, press the green TALK (flash) key, then dial the extension, then press "END" to complete the transfer.

Verify coverage (How to put a handset in RSSI test mode): Press Menu, # # # # *, then select "BER test".

Use the arrow up/down button to scroll to "RSSI testing". A -70dBm or stronger signal is excellent. -80dBm is good. -85dBm or weaker is not ideal. The location of the base or antenna of the base should be where the signal is at least -90dBm or better. See the "RSSI instruction sheet" for more information.

How to enable/disable headset broadcast control: If plugging in a headset causes the phone to go into broadcast mode, you will need to turn this feature OFF (because this feature is only available with call control type headsets). To disable, press menu, # # # # * then using the arrow up/down button select “headset button”. Turn it OFF and SAVE. If you have a call control type headset, enable this feature.

Installing multiple base stations at the same location: To avoid self-interference, separation of base antennas at least 30 feet or more for best performance.

How to check firmware version: Press Menu # # # # *, then using the arrow up/down key select “HS SW/HW Version” and/or “BS SW/HW version” to gather handset and base firmware version information.

How to find out the IP address of the base station: From a registered handset, press menu # # # # *, then using the arrow up/down key, select “BS SW/HW version”. Then scroll to see the IP address information.

Default “PIN” for handset: 0000. If you forgot the pin, see Ref. #4 (resetting a handset).

Ref #1 – Registering a handset: First power up the base station for at least 5 seconds. Next, locate the registration button near the DC-In power jack (labeled “REG”). Press and hold the REG button until you hear a song playing. Now grab the handset you want to register and press Menu 7, 1. The display will say “Registration complete” and the handset should get a new ID. Handset ID’s start at 10 and will go up to 19 (allowing up to 10 unique ID’s and handsets to be registered to the system). Note: You can also initiate base station registration remotely via the GUI by going to “TOOLS”, then “Reset/Reboot/Reg” and click the “REGISTER” button.

Ref #2 – Deregistering a handset: Same procedure as Ref #1, except press Menu 7, 2 instead of menu 7, 1. Keep in mind, deregistering a handset is not required to register a handset so if you have a handset that already has a registration ID but is not communicating with the base, go ahead and register it again (it will override any previous registration information with a new registration ID). If you need to bring back a specific handset ID number (a typical issue when getting back a handset after repair – see Ref #3.

Ref #3 – Recovering a handset ID (typical after having a phone repaired): All handsets returned for repair must be registered to the EnGenius repair dept. test base station. This causes your handset lose registration to your base station. Unless you deregistered the handset from your base station prior to sending it to EnGenius your base station still thinks that handset still exists. For example, if you have four handsets registered as ID 10, 11, 12, and 13 and you send handset 12 in for repair, when you reregister it, you will get ID 14 because it thinks 12 is in use. To get your handset registration back to ID 12 you must erase handset 12. From any handset press Menu, 6, then enter the PIN (default is 0000) then scroll up/down and select “Clear HS”. Now enter the two-digit handset ID you want to clear. In our example it would be handset ID12. Now ID12 is clear and available as the next registration ID. The base unit gives IDs to handsets in numerical order of free IDs. You can now go to Ref #1 (above) and go through the normal handset registration process.

Ref #4 – Resetting a handset: Press Menu, # # # # *, then scroll down to “System Reset”. Press “SELECT” and then press “Clear All” or “Keep Registration”. “Keep Registration” clears all settings except registration to the Base Station and “Clear All” clears everything including previous registration information.

Ref #5 – Resetting the base station: First power up the base station for at least 5 seconds. Locate the reset button next to the DC-In power jack. It's the very small button labeled "Reset". Now use a pen or paperclip to press and hold this button down until you hear it beep. Next power cycle the base station by simply removing power and then powering it back on. The base station is now reset (registration will now start at ID10 again). Default IP address is 192.168.1.156 with user name and password "admin". Note: You can also reset the base station remotely via the GUI by going to "TOOLS", then "Reset/Reboot/Reg" and click "Reboot".

Ref #6 – Assigning a handset to a group: Press Menu, 4 (Call Settings), then 2 (Group Setting), then choose "option" to "subscribe" to a group or groups.

NOTE: The FreeStyl-SIP system supports up to 7 definable groups and also the "all group" (which all handsets are part of). Also note, regarding inbound/outbound calling, grouping only allows a handset to receive incoming calls from more than one account. For outbound calls, the handset will still only use the SIP account the handset is assigned to.

Ref #7 – Turning up or down the handset microphone sensitivity: Press Menu, 5 (Phone Settings), then press 1 (Mic. Gain) and change accordingly.

Ref #8 – Increasing the front speaker gain (volume out): Press Menu, # # # # *, then scroll up/down to "Volume adjust". Then select "Receiver out". Ref#9 – Experiencing echo on calls: In the base station GUI, go to System>VoIP>Audio Settings, then under Codec/Hardware you will find Speaker Gain. Default is -8dB. Try changing it to -12dB to mitigate echo.

FreeStyl SIP FAQ

Q1: What is the default IP address, user name, and password?

A1: IP Address: 192.168.1.156, User Name: admin, Password: admin

Q2: What bundles/packages are available?

A2: System (Handset and Base), Handset only kits, and base only.

Q3: Can the FreeStyl 2 model work with the FreeStyl SIP model's equipment?

A3: No, the FreeStyl 2 is not compatible with the FreeStyl SIP equipment.

Q4: Can DuraFon SIP model's handset or base unit work with a FreeStyl SIP model?

A4: Yes, the DuraFon SIP is compatible with the FreeStyl SIP model. Note: the two model's handsets are compatible with either model's base unit, but batteries, chargers and other accessories are different.

Q5: Is the FreeStyl SIP a multi-base system?

A5: No. You can have several, separate base stations as long as you install them at least 30 feet apart to avoid interference with each other. Note: the 2- Way intercom and “Push-to-Talk” features would not work across handsets registered to different bases and calls can only be made on the handset’s specific base.

Q6: Is a repeater base available as an option for FreeStyl SIP?

A6: No.

Q7: Can a handset receive a phone call during a 2-Way intercom call conversation?

A7: No. It is important to note; no missed call notification will occur either. For this reason, SIP-to-SIP calling from one handset to the other is the preferred method.

Q8: Are there DuraWalkie handsets available for the FreeStyl SIP System?

A8: No.

Q9: Is the FreeStyl SIP field firmware upgradable?

A9: Base Stations can be, but handsets cannot.

Q10: Is there a log file available for call activity and history?

A10: No.

Q11: What is the maximum number of handsets that can be registered to the FreeStyl SIP system?

A11: 10.

Q12: From the Base Unit’s GUI, can I monitor or see if calls are in progress?

A12: No.

Q13: What frequency does the FreeStyl SIP use?

A13: 900MHz unlicensed ISM band (FHSS 902-928MHz).

Q14: Will the FreeStyl SIP work with Wi-Fi and is it a Wi-Fi phone?

A14: No. It uses its own 900MHz frequency and only works with its own matching FreeStyl SIP (or DuraFon SIP) base unit.

Q15: Where can basic set-up instructions be found?

A15: Refer to the FreeStyl SIP product webpage for a number of resources, including: user's manual, videos, guides and other important references.

Q16: What are advantages in using the FreeStyl SIP verses a standard FreeStyl 2 system?

A16: 1. It can plug in anywhere there is network connectivity. There is no need to run RJ11's from the IDF/MDF. 2. The FreeStyl SIP support a maximum of 4-talk paths for 4 concurrent calls. 3. The base station is field firmware upgradable. 4. Its GUI supports easy access to advanced audio adjustment parameters and configuration files can be saved.

Q17: Will the FreeStyl SIP interfere with Wi-Fi networks?

A17: No. Wi-Fi uses 2.4GHz and 5GHz, while the FreeStyl SIP uses 900MHz. However, it is always recommended to keep the DuraFon system at least 3-feet away from any other electronics.

Q18: How many simultaneous calls can a handset manage?

A18: Two. 1. To answer a second incoming SIP call from a different SIP extension (grouping): Press the "Switch" soft key. 2. To answer a second incoming call coming from the PSTN line: Press the "Switch" soft key. 3. To answer a second incoming call from a SIP extension while on a PSTN line call: Press the "Switch" soft key. 4. To answer a second incoming call from the same SIP extension, where call is coming from the PBX call waiting feature: Press the "TALK/Flash" key to answer and press the "Switch" soft key to switch between the two calls.

Q19: Can one handset have more than one SIP account associated to it?

A19: No. Each handset ID has its own SIP account, but you can utilize grouping to access more than one extension.

Q20: What VoIP codecs are supported?

A20: G.729 (v1.05), G.711a and G.711u3

Q21: Does the FreeStyl SIP handset support line appearances?

A21: No.

Q22: What's the maximum SIP account password character length?

A22: 32 characters

Q23: If two Freestyl SIP Handsets (associated with different FreeStyl SIP Base Stations) were used behind the same PBX, can one Handset call another?

A23: Yes. You can call via internal PBX extension-to-extension calling. You cannot however, 2-Way Intercom or Broadcast to the other Handset registered to a different FreeStyl SIP Base.

Q24: What is the system's range?

A24: Up to 25,000 sq. ft. with about 6-floors of penetration (hotel / office-building), 100,000 sq. ft. (retail store), and 10 acres (open area, like a farm).

Q25: Can the FreeStyl SIP Handset make conference calls?

A25: Yes, FreeStyl SIP can support two handsets and an outside phone call or a FreeStyl SIP handset and two outside calls.

Q26: What is the typical talk time and standby time?

A26: Typical talk time is 4-5 hours with standby up to 50 hours.

Q27: How do you make the handset automatically choose its SIP extension when dialing out?

A27: Note: Use the “up/down arrow” soft key to scroll: Press “Menu,” “4. Call Settings,” “3. Call Manager,” “Enter” Password: “0000” then press soft key “OK,” 2”. Outgoing,” then you can set it using 3 options: 1. Auto, 2. Manual, 3. Off. 1. Auto = will automatically choose the open line. 2. Manual = user will choose the outbound port. 3. Off = disable all outbound calls.

Q28: How many groups does the FreeStyl SIP support?

A28: 7. Groups allow a Handset to receive/make inbound calls from different SIP extensions (share). The maximum concurrent number of calls a handset can do is two (2). Note: You will need both an individual handset SIP extension AND another SIP extension set up for grouping. There is a separate document on FreeStyl SIP grouping available on the FreeStyl SIP product webpage.

Q29: Is the FreeStyl SIP Base Station PoE (Power-over-Ethernet) capable?

A29: No.

Q30: What is the Handset password?

A30: The default Handset password is 0000.

Q31: How can I find out the IP address of the FreeStyl SIP Base Station?

A31: From any registered handset: Press “Menu ####*”, scroll up/down to BS HW/SW Version and press: “select,” now scroll up/ down until you get to the IP address screen.

Q32: How do I transfer a call from an EnGenius handset to a different extension on my PBX phone system?

A32: For most PBX’s, press the “Flash” key while on an active phone call to initiate a call transfer. The green “TALK” key serves as the “Flash” key while on active phone calls. After pressing Flash, enter the extension you want to transfer the call to, then press END to complete the transfer.

Q33: How far can 2-way Intercom and Broadcast go (handset to handset communication)?

A33: Because these modes do not use the base station, range in 2-way intercom or broadcast mode is typically 50% less than for telephone calls. The difference can be even greater if an external antenna is used on the base station. Overall range will vary depending on environment.

Q34: Can I register handsets to multiple FreeStyl SIP base units?

A34: No. A handset can only be registered to one base station at a time.

Q35: What characters are allowed for SIP name, password, etc.?

A34: (1) SIP Number : a-zA-Z0-9$*()_+-= (2) Display Name: a-zA-Z0-9!@#$%^&*()_+-=,. (white space) (3) Password: a-zA-Z0-9!@#$%^*()_+-=, (white space) & (4) Auth Name: !#$%&'()*+,-./0-9:;<=>?@A-Z[]^_`a-z{|}~

MISC. QUESTIONS FROM CUSTOMERS

Q-1: How many users can the FreeStyl SIP support?

A-1: The FreeStyl SIP supports up to 10 handsets, with each handset having its own SIP client account. The base can handle a maximum of 4 concurrent calls.

Q-2: Are there any other model EnGenius systems that support SIP?

A-2: Yes, currently the FreeStyl SIP and DuraFon SIP, with more to come. Please check the EnGenius voice products webpages.

Q-3: Are the conversations secure and HIPAA compliant?

A-3: Yes and the FreeStyl SIP meets HIPAA compliancy.

Q-4: I have a customer using two FreeStyl SIP base units, with handsets on each; can these handsets communicate via digital two-way radio with each other?

A-4: Only the handsets registered to the same base unit. The other handsets would have to be “called” via an IP-PBX or SIP service extension or phone number.

Q-5: Does the FreeStyl SIP support external antennas or repeaters, like DECT phones?

A-5: The FreeStyl SIP base unit can utilize the optional indoor or outdoor external antennas. An antenna splitter (SN-ULTRA-AS) is also available to be able to run two antennas to the base unit.

Q-6: Where can I find information about the optional antennas and other accessories?

A-6: Accessories can be found on these webpages: https://www.engeniustech.com/phone-accessories.html

Q-7: Can a dealer update the firmware?

A-7: Yes, for the base station, No for the handset. Handset will require an RMA if firmware updating is needed.

Q-8: Can I handle multiple calls on one SIP handset?

A-8: Yes, with a maximum of 2. Maximum concurrent calls on the base is 4 (4 talk paths and 10 SIP accounts).

Q-9: Where can I learn about basic set-up information for hosted SIP providers?

A-9: The hosted provider general has basic set-up instructions and the FreeStyl SIP product page also has a number of provider set-up instructions via documents and / or videos.

Q-10: Can I adjust mic-gain settings for the handset?

A-10: Yes, via hidden menus: Menu ####* or Menu ****#.

Q-11: What would an Avaya IP Office need to work with the FreeStyl SIP?

A-11: A 3rd party IP license(s) are required on an Avaya IP Office with one each per handset. Here is a link on YouTube on setting this up: https://www.youtube.com/watch?v=2TKNrXT7V5I

Q-12: Where can I find out how to use many of the function with my 3rd party IP-PBX or Hosted SIP provider?

A-12: The best place to go for these type of instructions would be the maker of the IP-PBX or the hosted SIP provider. Let them know you are using a 3rd party device behind their system. Not all functions will be accessible with 3rd party devices.

RMA requests or support – go to support section on www.engeniustech.com or email support@engeniustech.com