DuraFon SIP Quick Reference Sheet

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

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-durafon-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. -90dBm or weaker is not ideal. Location of the base or antenna of the base should be where signal is at least -90dBm or better for all locations that need coverage. 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).

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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 DuraFon-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.

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DuraFon® 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 a DuraFon Pro handset work with a DuraFon SIP base station?

A3: No, the DuraFon-SIP is not compatible with any other EnGenius handsets or base stations.

Q4: Is the DuraFon SIP a multi-base system?

A4: 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.

Q5: Is a repeater base available as an option for DuraFon SIP?

A5: No.

Q6: Does the DuraFon SIP support TAP and text messaging?

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 to go along with the DuraFon SIP System?

A8: No.

Q9: Is the DuraFon 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 DuraFon SIP system?

A11: 10.

Q12: From the base Station GUI, can I monitor or see if calls are in progress?

A12: No.

Q13: What frequency does the DuraFon SIP use?

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

Q14: Will the DuraFon 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 DuraFon-SIP base station.

Q15: Is the DuraFon SIP system BroadSoft ™ certified?

A15: Yes. Make sure though the base has version 1.10.2 or higher.

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DuraFon SIP FAQ continued

Q16: What are advantages in using the DuraFon SIP verses a standard DuraFon 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. It has intercom/broadcast at the base station (like DuraFon 1x) (PRO does not).

3. Both DuraFon PRO and DuraFon SIP support a maximum of 4-talk paths, but unlike PRO, 10 unique DN (ext. numbers) can be associated to the base (with PRO, only 4 via 4 separate RJ11 connections per base).

4. The DuraFon SIP System is a network device with a unique IP address and can be remotely managed.

5. The base station is field firmware upgradable.

6. Its GUI supports easy access to advanced audio adjustment parameters and configuration files can be saved.

7. Also has (1) RJ11 analog port for fail over if "SIP server not available".

8. No need for a PBX analog (ATA) adapter / analog station card / voice gateway.

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

A17: No. Wi-Fi uses 2.4GHz and 5GHz, while the DuraFon 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 (v1.05 and later).

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 DuraFon SIP handset support line appearances?

A21: No.

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

A22: 32 characters (F/W v1.05 and later)

Q23: If two DuraFon SIP Handsets (associated with different DuraFon 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 DuraFon SIP Base.

Q24: What is the system's range?

A24: Up to 12-floors (in-building), 250,000 sq. ft. (warehouse), and 3,000 acres (open area).

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Q25: Can the DuraFon SIP Handset conference calls?

A25: **Device**	**Caller 2**	**Caller 3**	**Capable**
DuraFon SIP	DuraFon SIP	Outside line	YES
DuraFon SIP	PBX Desk-set	Outside line	YES
DuraFon SIP	Outside line	Outside line	YES
DuraFon SIP	DuraFon SIP	DuraFon SIP	YES
DuraFon SIP	DuraFon SIP	DuraFon-PSTN	YES

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, either SIP or PSTN.

2. Manual = user will choose the outbound port.

3. Off = disable all outbound calls.

Q28: How many groups does the DuraFon 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 DuraFon SIP grouping available by emailing support@engeniustech.com

Q29: Is the DuraFon 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 DuraFon 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 25% to 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 DuraFon-SIP base stations?

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{|}~

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MISC. QUESTIONS FROM CUSTOMERS...

Q-1: I need 5 cordless handsets for a client. I see what looks like a 1X. This will sit behind a PBX supporting SIP. Do I need to purchase just one base to accomplish this?
A-1: Yes, just one base station is needed. The DuraFon-SIP supports up to 10 handsets, with each handset having its own SIP client account. The base can handle a maximum of 4 calls at the same time.

Q-2: Are there any other model EnGenius systems that support SIP?
A-2: No.

Q-3: Are the conversations secure and HIPAA compliant?
A-3: Yes and the DuraFon SIP meets HIPAA compliancy.

Q-4: I have a customer with a DuraFon Pro and they now need connections in another building. Can the EnGenius SIP talk to the DuraFon Pro thru radio or just thru PBX lines?
A-4: Only through internal extension to extension calling on the PBX. The DuraFon Pro and DuraFon SIP are not compatible with each other.

Q-5: Does this unit support use of multiple external antennas with RF splitters (like we do with cell phone repeaters)?
A-5: The DuraFon SIP base station uses the same RF connector and antennas as the existing DuraFon product line, so the SN-ULTRA-AS antenna splitter will work and optional indoor SN-UL-AK20L-IND antenna and optional outdoor SN-UL-AK20L antenna will work as well.

Q-6: I need the specs on the new optional indoor antenna. Building is nearly completed.
A-6: Specs can be found here: https://www.engeniustech.com/engenius-products/phone-accessories-antenna-cable-kit-sn-ul-ak20l-ind/

Q-7: Can I as a dealer update the firmware ourselves?
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: Are there integration /application notes for tested phone systems? Will EnGenius be taking these through the manufactures interoperability labs?
A-9: Email support@engeniustech.com to find out if the DuraFon SIP has been tested or used with a particular IP-PBX or hosted provider.

Q-10: Can I adjust mic-gain settings for the handset (like from the secret menu on 1X/PRO versions)?
A-10: Yes via hidden menus: Menu ####* or Menu ****#.

Q-11: What would an Avaya IP Office need to work with the DuraFon 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: I have a question on functionality. Is there a guide that indicates how the Cisco features are used with the handset? Example, if our park code is *1020, would I simply press the talk key and then dial *1020 to park the call from the DuraFon handset? Most calls are retrieved from park in the warehouse and counter so I'm assuming we wouldn't park too many calls but just wanted to be sure we can park and retrieve easily. And to transfer a call, would I simply press the "talk" key and enter the 4-digit extension I am transferring to?
A-12: We do not have a Cisco specific guide however the process should be the same for any 3rd party SIP phone such as DuraFon SIP using the Cisco system. Regarding the process itself, (in the example, simply press TALK + dial *1020 to park), the answer is yes and also yes to regards to how to transfer a call. We do recommend leveraging our speed dial and/or phone book to make the process easier for common or heavily used dial strings. Another example would be overhead page, call park retrieve, and getting access to voicemail etc. Refer to the phone book in the manual for the process on that. If there are any issues, please contact our tech support and we will be glad to help.

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Q-13: We are connecting to an NEC SV8100 (would be the same as SV9100) and the only sip server setting it shows is for one account. The extension changes to what the PBX extension is so that the handset connects and works. With a 2nd handset registered with the 2nd extension the phone system won't register the extension. How do multiple extensions off of a PBX register to the base for use or even multiple talk paths off 3rd party SIP with the one base station?

A-13: In the NEC you have to allow for all extensions connecting to the same base unit duplication of the mac address.

Still having trouble? Email technical support at support@engeniustech.com

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