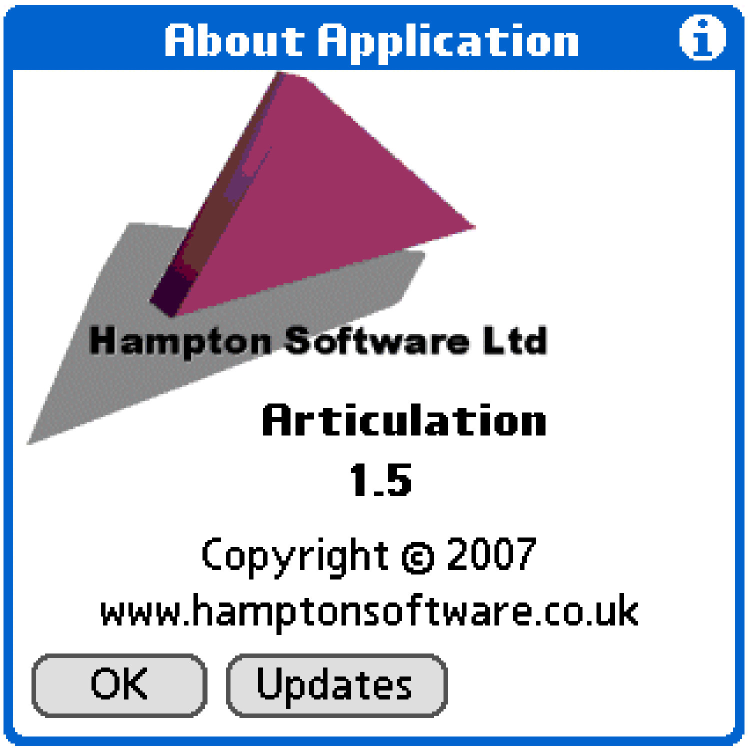
Articulation

For
Palm OS®



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1. Introduction

Articulation is an easy to use application that turns your Palm PDA into a VoIP Phone, enabling you to make and received telephone calls using the Internet. In order to use Articulation you need a Palm PDA that has the following:

- Speaker and Microphone
- Internet connectivity (through WiFi for example)
- Palm OS 5

If available for your particular PDA, headphones will reduce echo and improve the call quality; especially if your PDA supports headphones with a built in microphone.

You will also require an account with a VoIP service provider. There are many of these available on the Internet, and many allow you to sign up for free in order to test the service.

Before purchasing Articulation it is **essential** that you test the software with your Internet connection and VoIP provider. Internet connection speeds and other network factors may mean that you cannot make/receive calls or the voice quality may be poor (although it may be possible to fix the latter – see the Troubleshooting section later in the manual).

2. Getting Started

2.1. Installation

Installation of Articulation is straight forward, simply double click on the *Articulation.prc* file downloaded from your supplier and install as normal using Hot Sync.

If you hardware or PDA configuration does not meet the requirements for Articulation you will see an error explaining the problem and you will be returned to the Application launcher.

2.2. First Call

When you first run Articulation it will display a welcome screen that provides an *Echo Test* call option to allow you to try your Internet connection with VoIP. If this call does not work, check with the troubleshooting section regarding possible problems.

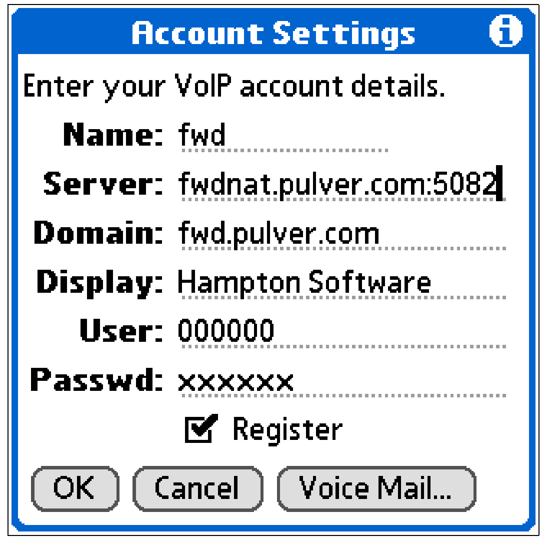
From the main screen, the *Call* menu contains two test number which may be used to test your network setup before you select a service provider.

The *Info test call* option will place a call to an information service (in the US) and allow you to listen and interact with the service. This service uses voice recognition – if the service is continually telling you that it does not understand, try turning the microphone volume down from the *Options->Phone...* menu option. The microphones on some PDAs are very sensitive.

The *Echo test call* (same as that used by the welcome screen) will place a call to an echo test server (again in the US); this call will enable you to hear how you sound and will give you an idea of the delay in speech for your network. If you are not located in the US, the sound quality may vary – in this case you will need to find a local VoIP provider.

If you are not using headphones you will hear feedback effects with the *Echo test*. The echo cancellation feature is not enabled when using the *Echo test call* since it stops the echo test from working!

2.3. VoIP Provider



Once you have established that your network connection works for VoIP, you should sign up with a provider. You should test Articulation with your particular provider before committing to use them since it is still possible that Articulation may not be compatible with your provider. The details provided to you by your VoIP provider may be entered under the *Account...* option of the *Options* menu.
A list of the popular VoIP providers and their settings is given at the end of this manual.

As an example, if you sign up with http://www.freeworlddialup.com you will be provided with a user number and a password and a proxy server and domain (*fwdnat.pulver.com:5082* and *fwd.pulver.com* in this example). These settings should be entered as shown in the screen shot on the right.

Different providers may use slightly different terminology; server, registrar, proxy and outbound proxy relate to the server, within Articulation, and domain and realm relate to the domain, within Articulation. You may be provided with more information than required – for example a value for a registrar and an outbound proxy; simply try either to determine which one works.

If you do get the values wrong, simply select *Account...* from the *Options* menu, or tap the connection icon at the top of the screen, and try again. When the account settings are correct and Articulation has registered with your VoIP provider a green connection icon is shown at the top left of the screen (during connection this may flash red and green, and when registration fails the icon shows an unconnected red plug).

Once the green connected icon is showing you are ready to make and receive calls with your service provider.

> **Treo only:** As well as the VoIP connection icon, the bluetooth, cell phone and battery icons are displayed on the main screen.

2.4. Multiple VoIP accounts

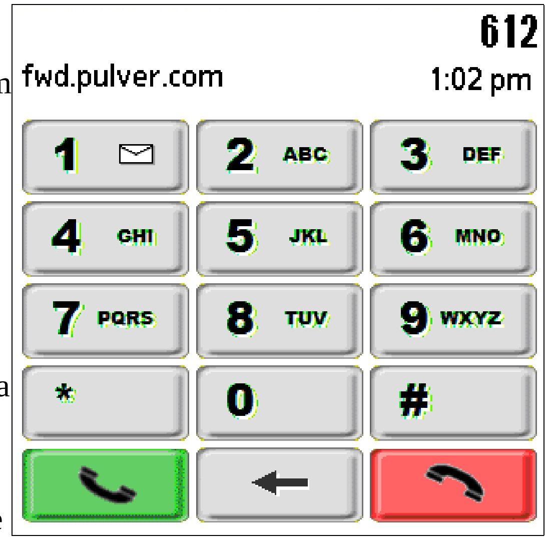
You can subscribe to as many VoIP providers and accounts as you like and have access to all of them through Articulation. This may be useful for connecting to friends who are subscribed to different VoIP providers, and for selecting the best provider for calling particular countries.

A pop up menu of the providers is given at the top of the screen. Before dialing, select the account you wish to dial from. New providers and/or accounts can be added from the New Account... option on this menu.

When you select an account using this menu, the last status message received for that account will be shown. When there is a problem with an account the name will be preceded by an x in the menu, and selecting that provider will display the problem.

To modify the account details for a particular account, select the account in the menu before selecting the Options->Account... menu item (or tapping the connection icon). If you change the account name, you will be asked if you wish to create a new account with the details or replace the old account – this is useful for copying settings when creating new accounts.

2.5. Making Calls



Many VoIP providers have a free number you can call for testing we recommend calling this number first to determine the call quality from your network connection and provider.

The example on the right shows the *Echo test* number from Free World Dial up being called – this is a good number to start with since it allows you to hear what you sound like to the other end.

When testing with an echo test number you can hear the delay (called latency) introduced by the VoIP components and your particular network configuration. Do not forget, however, that you are hearing a slightly worse than normal delay (although not quite twice) since the voice must go to the other end and back.

If you experience voice quality or other problems during your call see the Troubleshooting section at the end of this manual for possible resolutions.

2.5.1. Dialing

You can dial a contact using several methods:

1. Simply dial the number required on the screen
2. Enter the number (or name) to dial using graffiti or a PDA keyboard. If you wish to dial a SIP URL simply enter the URL without the preceding *sip:* (for example, *613@fwd.pulver.com* calls the echo test number in this example)
3. Use the *Lookup* button at the bottom of the screen to select a number from your contacts list
4. Use the *Recall* button to recall the last number dialed (at the bottom of the screen when available).
5. Use a third party dialing application (e.g. Palm dialer) to start the call.

The number to dial is shown at the top of the screen – once you are ready to proceed with the call press the green dial button in the bottom left hand corner of the screen.

Progress of the call will be shown on the screen – and, when appropriate, tones will be heard through the PDA speaker. If the number dialed cannot be reached (if it is busy for example) this will be shown and you will be returned to the normal dialing screen.

By default, Articulation registers as the dialing application for the Palm dialer – if this is not required you can disable this feature from the *Options->Settings...* menu item.

> **Treo only:** Articulation does not register as the dialer application, by default, leaving the regular cell phone as the main dialing application.

2.5.2. Speaker Phone

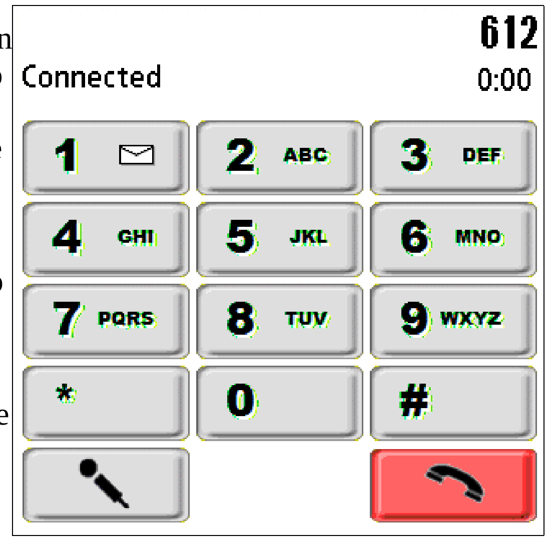
If you intend using your PDA in a speaker phone type mode (i.e. Without a headset) then it is advisable to switch *Echo Cancellation* on otherwise the remote party may hear very bad echo effects.

The *Echo Cancellation* can be enabled from the *Phone...* form from the *Options* menu.

Articulation uses a very simple type of echo cancellation – it detects voice from the remote party and mutes the microphone on the PDA while there is voice incoming from the remote party. In very noisy environments or on a noisy line it may incorrectly detect voice and mute the microphone producing a stuttering effect at the remote end. For this reason we recommend that a headset is used.

As an additional note, calling *Echo test* numbers while using the PDA as a speaker phone may cause very bad feedback effects (even with echo cancellation switched on).

2.5.3. In Call Options



While in conversation the screen will show the dial pad, a *Mute* button and an *End* button. The dial pad may be used to send DTMF digits to the remote end (for example, for entering options in a menu). You should note that DTMF digits you enter are not encrypted and may be intercepted en-route (in the same way that emails may be intercepted) so care should be taken when entering sensitive information.

The *Mute* button enables you to cut off the microphone at your end so that the remote party will not hear you; note that you will still be able to hear the remote party.

The *End* button allows you to finish the call and hang up – you will be returned to the normal dialing screen. The call may also be ended by pressing the rightmost hardware button or power off button on your PDA (this is, the end call button for Treo devices and the Notepad button for Tungsten devices).

The example screen shot on the right shows the screen during a call; the elapsed time for the call is shown below the dialed number.

To increase or decrease volume during a call, the navigate rocker up and down may be used (scroll up/down on some devices).

Treo only: The volume button on the side of the device will work during a call.

A *Spkr* button will be available to allow the Treo to work as a Speaker Phone. Normally calls through a Treo will be heard through the phone receiver speaker.

2.6. *Incoming Calls*

In order to check that incoming calls work correctly it is advised that you make a test inbound call as soon as possible. In order to test the incoming call, you either need another VoIP phone subscribed to the same VoIP provider or a DID (direct inward dial) number that is used to access your VoIP phone from the normal telephone network – many VoIP providers will provide you with a DID (often this is the same as the user name).

If you are using Free World Dial up it is possible to schedule a call to your VoIP phone through the *Call Me* service on their web site. Simply login to your Free World Dial up account and select the *Call Me* link.

When an incoming call arrives an alert will be displayed on the screen, this includes the identity of the caller (if known). You have the option to either *answer* the call or to *ignore* it – if you select *ignore* the caller will hear busy tone. The left most hard button on your PDA (answer call button on a Treo) can also be used to answer the call, and the right most or power button to ignore the call.

If you answer the call, the display will switch to the normal call display as you used with the outgoing call (see the In Call Options section above). As before, problems with the call may be resolved by referring to the Troubleshooting section; the call may be ended by pressing the red *End* button in the bottom right corner of the screen.

Note that Articulation does not support call waiting – if another incoming call arrives while you are busy on a call, the caller will hear busy tone (or may be directed to voice mail depending upon your VoIP provider).

Treo only: If you are using a Treo with a headset, the answer button on the headset also allows you to answer the call.

The sound & alert preferences for the phone application will also be used for incoming calls to Articulation.

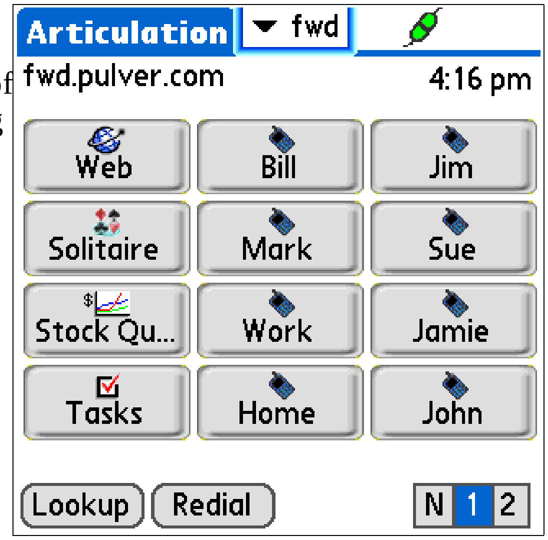
2.7. *Incoming Calls while in other applications*

Articulation can operate in the background, so that incoming calls may be received while you are using other applications. Normally, when you exit Articulation, you will be asked if you would like it to run in the background. This behavior can be modified from the Options->Settings... menu item, allowing Articulation to always, never or prompt for working in the background.

Also from the Options->Settings... menu item, you can set Articulation to provide an alert if network connection is lost while running in background mode. This alert will be canceled if Articulation re-establishes the network connection.

If you reset or hot sync your PDA you must run Articulation again in order to place it in the background. Note that if your network connection is lost while Articulation is running in the background you will not receive calls. Articulation will attempt to reconnect once the network connection is restored.

3. Speed Buttons



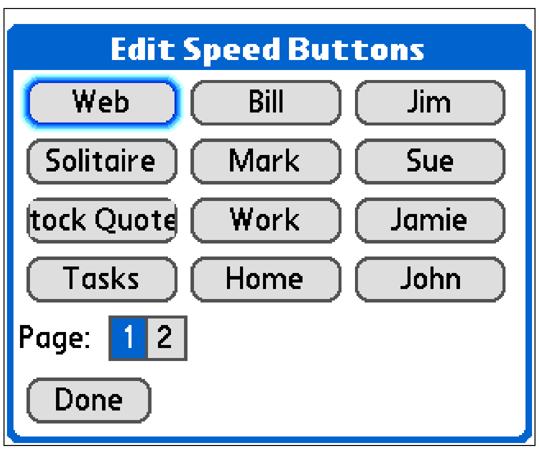
Speed buttons allow you to quickly access commonly called numbers or applications direct from Articulation. Articulation has two pages of twelve speed buttons; the page of speed buttons is selected by tapping the appropriate push button in the bottom right of the screen. The speed button pages are identified by 1 and 2 and the numeric keypad by N.

The display to the right shows an example of a speed dial page, with page 1 selected.

Each speed button can contain either a number to dial or an application to launch. If a button has not been assigned it will be shown as blank – tapping a blank button will allow you to edit the content of the button.

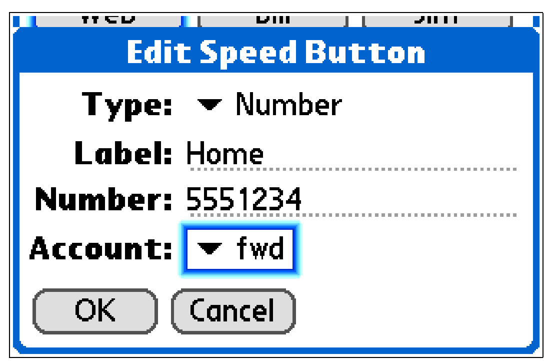
Tapping a number speed button will cause the number to be dialed immediately. Tapping an application speed button will launch that application immediately.

3.1. Editing Speed Buttons



The Buttons... option from the Options menu may be used to edit the buttons. This option will display the layout of the buttons and a selector to switch between pages. An example of this is shown on the left.

Tapping on one of the buttons will allow you to edit the button (as shown below). For number speed buttons you can enter the number to dial and the VoIP account to dial from. If you leave the Account field blank, the currently selected VoIP account will be used.



4. Voice Mail

Articulation contains a number of features to assist you with voice mail. Firstly it will notify you when voice mail is present, and then by pressing and holding the *1* button on the dial pad it will call your voice mail service so you can pick up your messages.

Voice mail is configured from the *Voice mail* button on the account set up form. An example of the Voice mail settings form is shown on the right.

4.1. Voice Mail Notifications

Unfortunately VoIP standards for voice mail notifications are very new and are not supported by all service providers and vendors. There are two main types of voice mail notification, both of which are supported by Articulation. One of the types, however, may cause an error message to be displayed when you first connect to your service provider. This error does not mean that you will not receive voice mail notification, only that it is possible you will not! The only way to tell is to leave yourself a voice mail message and run Articulation to see if it indicates you have voice mail (you may need to wait a minute or two after it has connected to your service provider).

Where you have multiple accounts, and voice mail notification arrives, an envelope icon will be displayed against the VoIP account that has voice mail on the accounts pull down menu. In some rare cases, it is possible for the voice mail to be assigned to the wrong account (due to insufficient information in the notification message!).

Some service providers may require you to use a voice mail URI to check for voice mail notifications. This is configured by entering this URI into the *Account* field on the voice mail configuration form.

If *Alert me of new messages* is checked, and Articulation is running in the background, an alert notification will be given when new voice mail is detected.

When running Articulation and voice mail is available, an envelope icon will flash in the top right corner and an alarm sound will be made.

4.2. Accessing Voice Mail

Articulation can be configured so that pressing and holding the *1* button (with the mail icon) will call your voice mail service. If your service provider provides the correct information within the voice mail notification, Articulation will automatically configure the number to dial to retrieve voice mail; otherwise, enter the number in the *Access* field on the Voice Mail settings form.

You often need to dial digits to enter an account number or password before accessing your voice mail, you can enter this in the *Post dial* field. If you enter the character *p* this will be replaced by a pause of 1 second. In the example screen shot above, once the voice mail has connected it will pause for 4 seconds dial seven zeros, pause for 4 more seconds and dial four zeros. Some trial and error is required here to get the right sequence for your particular voice mail service.

5. Registering

Articulation places a 45 second limit on call length until a registration key has been purchased. To purchase a key please visit the web site from which you downloaded Articulation. Your Hot Sync id is required to complete the purchase.

Once you have been sent a registration key, select the *Register Now* option from the *Options* menu and enter the key.

6. Support & Updates

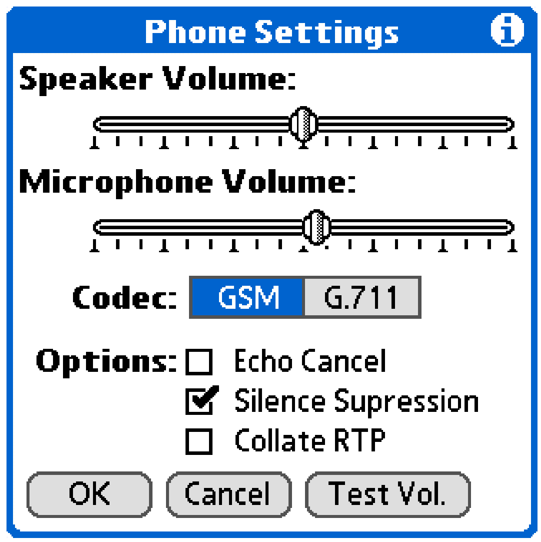
Selecting the *About...* option from the *Options* menu will give you instructions on obtaining support (select the tips icon in the top right). Please ensure you have read the troubleshooting section in this manual and on our website below before contacting us.

From the *About...* option you can also check if there are any program updates available by selecting the *Updates* button. Instructions on obtaining the update will be provided.

7. Troubleshooting

Voice over IP is a relatively new technology and as such there are a number of potential problems that may be encountered. This section will attempt to address the problems that may be encountered.

Many of the problems described below may be resolved using the *Phone...* form which is available from the *Options* menu. An example screen shot is shown to the right.



7.1. No Voice/Sound in one or both directions

There are a number of possible causes for no voice/sound:

1. Volume setting. The speaker/microphone volume may be turned down – the volume can be increased from the *Options->Phone...* form or by using the navigate up/down rocker (or scroll up/down).

2. Silence suppression is switched on. Silence suppression saves network bandwidth by not transmitting when you are not speaking – it is possible that Articulation has incorrectly determined that you are not speaking and therefore is not transmitting. To resolve this un-check *Silence Suppression* from the *Options->Phone...* form.

3. Echo cancellation is switched on. Echo cancellation works by muting the microphone when voice is detected from the remote party – if the remote party is in a noisy environment it is possible that the echo cancellation is muting the microphone. To resolve this un-check *Echo Cancellation* from the *Options->Phone...* form. Note that you should use a headset in this case otherwise the remote party may hear echo.

4. You have an incompatible NAT router. Some routers/gateways that connect your PDA to the Internet are of a type that do not allow traversal of VoIP voice streams without the correct configuration. Your VoIP provider may be able to provide an alternative server setting that will allow the service to work;

please check with your VoIP provider.

As an example, if the free world dial up services does not work using *fwd.pulver.com* as the server, try *fwdnat.pulver.com:5082* in the server setting (leave the domain as *fwd.pulver.com*). This alternative server is configured to work with all types of NAT router.

7.2. Poor /choppy voice quality

Poor voice quality (usually heard as stuttering or breaks in the speech) is most often caused by insufficient network bandwidth. That is, there just isn't enough capacity on your Internet connection to handle a VoIP call. This could just be a temporary effect (for example, someone is performing a large download on the same Internet connection) in which case the voice quality should improve.

There are a number of steps that can be used to minimize these problems:

1. Silence suppression. This option stops your PDA from transmitting during the periods in which you are silent – on average this reduces the transmitting bandwidth use by 50%. *Silence Suppression* can be enabled by checking the option on the *Options->Phone...* form.
2. Codec selection. Articulation supports two codecs G.711 and GSM; these are methods for encoding the voice while it is transmitted over the Internet/network. In the non-VoIP world, G.711 is the encoding used by the normal telephone network and GSM is most often used on the mobile telephone network. While G.711 provides better quality voice, it uses much more bandwidth than GSM (almost four times more). The benefits of the reduced bandwidth usage often provide a better call quality with GSM. To modify the codec selection use the *Options->Phone...* form. With both GSM and G711 selected Articulation and your VoIP provider will attempt to negotiate the encoding to use.

   Note: Not all VoIP providers support GSM (they should all support G.711) so deselecting G.711 may not work.

3. Collate RTP. Normally, the voice data is transmitted in 20ms packets; the collate RTP option transmits the voice in larger packets (up to 60ms) in order to reduce the bandwidth used by packet overhead. This is not be compatible with all voice providers and where it is not compatible, it will result in very poor voice quality. For GSM this option can save almost 5kbps of bandwidth. *Collate RTP* can be enabled by checking the option on the *Options->Phone...* form.

Codec Selector

In order to determine the best codec for your network you can use the following table:

Network Type & Bandwidth	***G.711 Codec (~80kbps)***	***GSM Codec (~22kbps)***
Voice Modem 56kbps down, ~33kbps up	Not possible	Although possible it is unlikely that this would work well due to other technical problems.
Broadband via WiFi/Bluetooth 128kbps down, 128kbps up	Although possible, it is quite likely that other network traffic may cause some interference in speech.	Best solution for this speed

Network Type & Bandwidth	*G.711 Codec (~80kbps)*	*GSM Codec (~22kbps)*
Broadband via WiFi/Bluetooth >256kbps down, >256kbps up	Should work fine. It is possible that other network traffic may cause some interference in speech.	Best solution for this speed
GPRS* 115kbps	Although possible, it is quite likely that other network traffic may cause some interference in speech.	Best solution for this speed
EDGE* 384kbps	Although possible, it is quite likely that other network traffic may cause some interference in speech.	Best solution for this speed
EVDO* 400kbps	Should work fine	Best solution for this speed

* **Important Note:** Wireless communication using the mobile telephone network often involves cost based on the amount of bandwidth used. A 3 minute 10 second GSM call can use up to 1Mb of data (for G.711, it takes only 51 seconds to use 1Mb of data). This should be taken into account when planning call costs; for example, a plan which charges $10 per Mb call costs will be over $3 per minute. Obviously a flat rate with unlimited data bandwidth would be more cost effective.

7.3. There is a long delay in the speech

You may experience bad delay (known as latency) between you (or the remote party) speaking and the other end hearing the voice. This will be due to one of the following problems:

- Too far from your VoIP provider – there is a long transmission delay between you and your VoIP provider. This may cause delay, especially if your VoIP provider is in another country or on another continent!
- Insufficient bandwidth – if your connection to the Internet is fully loaded your voice transmission may be queued awaiting transmission at your router/gateway.
- Packet loss – Poor network reception may cause packets to be lost, this can cause VoIP phones to increase delays in an attempt to compensate for packet loss.
- Large/incorrect jitter settings at the remote phone (refer to the documentation for the remote phone).

Some of these problems may be resolved by following the procedure described under the Poor Voice Quality section above.

7.4. I do not receive incoming calls

In order for incoming calls to work, your PDA must be permanently connected to the Internet through your wireless connection; also the Articulation application must be running (or running in background mode).

You may need to adjust the network settings on your PDA so that the network will not disconnect when it is idle. Care should be taken when leaving your PDA available for incoming calls as this will use your battery charge much faster than normal – it is recommended that you leave your PDA in its charging cradle when possible.

8. VoIP Providers and Settings

Location	Provider	Server	Domain
US	InPhonex	sip.inphonex.com	inphonex.com
US / Canada	StanaPhone	sip.stanaphone.com	sip.stanaphone.com
US	SIPphone	proxy01.sipphone.com	proxy01.sipphone.com
US	Vbuzzer This service may work with VoIP blocked cell phone data plans as it uses a non-blocked port (80).	vbuzzer.com:80	vbuzzer.com
US	Free World Dialup	fwd.pulver.com	fwd.pulver.com
US	FonoSIP	fonosip.com	fonosip.com
US	iptel.org	sip.iptel.org	iptel.org
US	Vonage A Vonage SoftPhone account is required to use this service.	sphone.vopr.vonage.net	sphone.vopr.vonage.net
UK	Babble	sip.babble.net	babble.net
UK	sipgate	sipgate.co.uk	sipgate.co.uk
UK	Draytel.org	nat.draytel.org:5065	draytel.org
UK / Europe	VoIPtalk	nat.voiptalk.org:5065	voiptalk.org
Australia / New Zealand	BBPGlobal	sip1.bbpglobal.com:5190	sip1.bbpglobal.com:5190
Australia	Firefly / Freshtel	cts-au.freshtel.net	freshtel.net
New Zealand	Slingshot iTalk	akl.italk.co.nz	akl.italk.co.nz
France	Freephonie This service supports G.711 only	freephonie.net	freephonie.net
Germany	VoIP Cheap	sip.voipcheap.com	voipcheap.com
US / Argentina	brujula.net	voip.brujula.net	voip.brujula.net