

Do you have a gut feeling that Voice over IP (VoIP) is about to find its way into your organisation? Are you being badgered by PABX and/or Networking equipment vendors to upgrade to VoIP enabled devices? Are you totally baffled by the implementation choices available? If you can identify with any of these questions, then this article is for you.

Three different generic implementation models are presented, and although they are not exhaustively comprehensive, they will give you a framework around which you can ask questions to prospective vendors.

The generic model always starts the same way - a dual network of WAN and PABX/PSTN, and finishes with a single WAN with Softswitch (or Soft PABX) control.



Figure 1 - Generic "dual network" model

So let's look at the generic starting point - we have a dual network with both a PSTN and a WAN. For simplicity's sake, I have purposefully NOT specified the type of WAN, whether it be leased lines, frame relay or VPNs.

Although this may have an influence on the VoIP equipment you buy, it doesn't change the VoIP part of the implementation strategy greatly.

The Central Office

PABX with E1 (or PRI or T1) connections to the PSTN.

There may be leased trunk lines, tie-lines and any other variety of connections.

There is also a router that connects the CO to the corporate WAN. If you want to eventually use VoIP beyond your own corporation, then this WAN will need Internet access, but for now let's keep it generic.

When voice traffic flows between the CO and any of the branch offices, the call goes directly to the PSTN, as does any external call and all incoming calls. Data traffic is completely separated from the PSTN with the exception of dial-up users as shown in the Small Office.

The Regional Office

The regional office is an exact model of the Central Office, only smaller.

PABX and router provide PSTN and WAN connectivity respectively.

Voice traffic between Regional and Branch offices may be via tie-lines to the central office, or directly to the PSTN via ISDN, E1/T1 or even multiple analogue lines.

The WAN is much more likely to handle inter-office traffic via the Central Office, and inter-branch office traffic via the regional office in a typical hierarchical design.

In other words, although regional/branch offices may call each other directly via telephone, any data traffic between these offices is likely to go via the Central Office. This data model may not end up being the best model for inter-office VoIP voice data traffic.

The Branch Office

Much the same as a Regional office, but this time I have explicitly included a Key System as the PSTN connectivity, because the chances are that if you have a Key System, it won't be an easy upgrade to VoIP.

Modern Key Systems are effectively small PABXs. WAN connectivity is as for a Regional Office, perhaps with less capacity.

The Small Office (or Home Office)

Although many Small Offices or Home Offices (SOHO) have upgraded to ADSL, I have left the generic SOHO as dial-up, just to emphasise the need for upgrade if this office is to participate in VoIP. Yes, VoIP can be done over dial-up, but making a telephone call so you can make another telephone call kind of defeats the purpose.

Besides, the quality you are likely to get is less than business quality.

[Migration Models](#)

So you've

decided that you might do the VoIP migration, but my networking vendor is telling me to do one thing while my PABX vendor has a totally different view. Are you totally confused about your options? To help you, I'll discuss three possible migration strategies. None is completely comprehensive, they are simply plausible scenarios, and I don't even get into the discussion of which signalling protocol you should use which is another influencing factor. In my models I have endeavoured to provide a staged approach that introduces WAN IP trunks as the first stage, followed by IP handsets at a later date. Some implementations work the other way, add IP handsets first then add the WAN links later. This is perfectly valid, but at the time of writing and the intended audience, the greatest VoIP savings can be made by replacing inter-office trunks.

I've called these models:

- 1) The Softswitch (Cisco Call Manager) model
- 2) The Incumbent PABX vendor model
- 3) The IP softPABX (Asterisk) model

Without arguing semantics (a PABX is a softswitch too!), the **softswitch model** is based largely on a possible Cisco Call Manager migration model. If I were to substitute SIP Proxy or H.323 Gatekeeper or MGCP Call Control Agent for the Cisco Call Manager, the model would mostly be the same.

The essential feature of the Softswitch model is that you need PSTN gateways for at least part of the migration, and these gateways will have to be able to be IP enabled and support one of the IP signalling protocols (H.323, SIP, MGCP, H.248). You will have to either get (Cisco) voice enabled routers, or IP functionality for your PABX. I've drawn this model based on taking the voice enabled routers path. I chose this because I figured that if you were going to upgrade you PABX to IP, you would probably be heading down the Incumbent PABX vendors model. The **Incumbent PABX vendors model** is quite different, because PABX vendors (not surprisingly) think that the PABX should serve the softswitch function, so the whole migration strategy changes. PSTN connectivity does not need to change, which is the greatest strength of this model. You continue to work with the vendor you have come to know and love over time, and support services are likely to continue as before. This model has many advantages for those who want to tread softly and causes the least disruption to the status quo if your existing equipment is easily upgradeable but be wary. You may find that your existing equipment needs to be replaced, in which case you might THINK you are following the Incumbent PABX vendor model, but essentially following the IP softPABX model. The **IP softPABX model** differs from

the Incumbent PABX vendor model in that you are **replacing** your existing PABX with a new-fangled IP version rather than upgrading. This may be because you've realised that your existing PABX equipment is so outdated that you need to replace it, or because you are trying to save thousands of dollars by using an open source PABX ♦ Asterisk.

A fourth model exists too that is only just starting to become possible. I call it **The no PABX model** because with the arrival of the VSP ♦ Voice Service Provider (or VoIP Service Provider, aka Telephone Service Provider ♦ TSP) you can cut over directly from PABX to no PABX in one fell swoop. This model may well become the *status quo* in the near future.

[The Softswitch \(Cisco Call Manager\) model](#)

I have described 7 stages in this migration path. It is assumed that the eventual aim is to completely replace all telephone handsets with IP models in all offices ♦ even if not for some time. In any case, you will at least need to consider the possible bandwidth demands such a migration would have.

Also assumed is that you have done your homework particularly with bandwidth and electrical power consumption calculations. Both these topics deserve discussion in their own right, but I don't cover them here.

[Step 1](#)

♦ Add voice enabled routers & upgrade WAN links

No matter what model you follow, the chances are that you will need to upgrade some, if not all, your WAN equipment to cater for the upgraded link speeds you've already calculated, and to implement the kind of QoS (Quality of Service) that you know that your VoIP network demands. The softswitch approach requires that you have IP controlled gateways, so Cisco routers do make an excellent choice. Router and gateway built into one device has been one of the driving design paradigms for the newest ♦ Integrated Services Router ♦ models that Cisco is so successfully flogging at the moment.




Figure 2 ♦ Add voice-enabled routers and additional bandwidth

The SOHO will need at least 128kbs bandwidth to be safely able to carry on a voice conversation and have data connectivity at the same time, so ADSL (if available) makes an excellent choice. If ADSL is not available, then you could use ISDN BRI, or a leased line. In places where the leased line is the only option, prices for a 128Kbs connection may be prohibitive (hello to my Papua New Guinean friends) and in this case it might be feasible to carry one or two compressed voice channels on your 64kbs leased line.

[Step 2](#)

- Add PSTN connections to Router

Now that you have your voice enabled routers, you will want to connect them to the PSTN for testing purposes. This means that for some time you will need to have dual circuits, the ♦old♦ ones for the PABX and the ♦new♦ ones for the router. Since you have calculated all your traffic statistics using the Erlang method, you may find that you need fewer PSTN trunks, because in the new model, most of your inter-office traffic will traverse the WAN. After all, this is probably what we are trying to achieve.

Many people will say this is also the point at which you add the softswitch, or Cisco Call Manager. I have purposefully left that to a later stage to emphasis the point that (particularly for smaller installations) you can

replace internal PSTN trunks using Cisco routers alone ♦ you don♦t need to buy the softswitch yet. You can program all the dial-plans and call routing rules you require on the Cisco routers for quite a few destinations before it becomes unmanageable.</p> <p

class="MsoBodyText"></p> <p

class="MsoBodyText">Figure 3 ♦ Add voice trunks to voice enabled

routers.</p> <p class="MsoBodyText">During this stage you will also need at least some

testing equipment to make sure your voice circuits and dial plans are working properly.

You should also be testing fail-over, making sure that if capacity is not available on the

WAN, then additional calls are diverted to the PSTN. When they are, you can

progress to cutting the PABX lines over to the router.</p> <h3>Step 3 ♦ Move PABX

trunks over to the IP Voice Gateway (router)</h3> <p class="MsoBodyText">This is a

crucial stage in the development. This is the point at which you cut the umbilical

chord from the PABX (or Key System) to the PSTN. You will probably need as

much capacity between the PABX and the router as you previously had between the PABX and

the PSTN given all users in this model still access all numbers via the PABX. This is also the

time when you may have to re-program the PABX as well, which makes it a particularly

vulnerable time, and is a major selling point for the Incumbent PABX vendor camp who say this

is all unnecessary and you shouldn♦t be using a router to do the job of a PABX.</p> <p

class="MsoBodyText">The major step you have taken is to reduce the number of

PSTN trunks you need, because the IP network is handling inter-office traffic.</p> <p

class="MsoBodyText">Another critical consideration at this point is how to control access to the

WAN so that a) not too many calls try and use the WAN at once and b) to fail over to the PSTN

should the WAN fail. In this scenario so far, we have used Cisco routers as our

combined PSTN/WAN gateways. This has some definite advantages as it is not

too difficult to program each router to allow a limited bandwidth allocation for voice, and use the

PSTN as a 2nd choice, but this is not a very scaleable solution.</p> <p

class="MsoCaption">
Figure 4 ♦ Divert PABX trunks to the gateway (router).</p> <p

class="MsoBodyText">In my model this can be the final step (for now). You

don♦t need to have VoIP phones just yet, but you have the ability to add them if you want

to. If your plan is to add many IP phones immediately, then you can reduce the

risk greatly by maintaining dual connections to the PSTN and diverting just a few lines to the

voice gateway, or even skipping this step altogether. In this scenario, PABX

users use the PSTN directly (as before) for all external calls, while the new IP phones access

the PSTN via the gateway. If this is the approach you take, you will need to

have already implemented the next step of adding the softswitch.</p> <h3>Step 4 ♦ Add the

softswitch ♦ Cisco Call Manager (or SIP server or♦)</h3> <p

class="MsoBodyText">This step could be taken at any stage really. Adding the

softswitch does nothing until you program your devices to communicate via it.

But the programming of the softswitch is such an important task that it deserves its own

step.</p> <p class="MsoBodyText">By now you will have decided on what protocols (H.323,

SIP, Skinny etc) you want to use in your network. Often the softswitch choice is

tied to the type of phones you decide upon, and the type of phones you decide upon determines the protocols you use, and the protocols you use will dictate the type of softswitch you choose and so on around in circles. Getting the right mix is not necessarily easy, so when Cisco come along and say, "All our stuff works together nicely" it's a tempting single vendor choice.

 Figure 5: Choose and install your softswitch.

In this plan, I have included a single softswitch or Cisco Call Manager (CCM) at the central office. There are many scenarios that you could model with multiple CCMs, some of them can be found on Cisco's web pages [here](http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_implementation_design_guide_book09186a00805fdb7b.html), and at the very least you should be implementing a dual-server approach for redundancy. Again, Cisco has a very structured approach that is to be recommended. If you can afford the initial outlay. Other choices of softswitch include the open source [SIP Express Router](http://www.iptel.org/ser/), Brekeke's [OnDO SIP Server](http://www.brekeke.com/products/products_sip.php), Open H.323's [Gnu Gatekeeper](http://www.gnugk.org/) and many voice vendors such as Nortel, Avaya etc have SIP &/or H.323 softswitches as well as IP enabled PABXs.

Call Admission Control (CAC)

A major consideration when choosing your softswitch should be: How are calls handled when there is not enough bandwidth on the WAN either because of failure or because other calls are using all the available bandwidth?

In other words, how do we control access to the WAN? In the PABX days, access to the PSTN was simply a function of how many physical lines you had. But in the IP model, there is no physical limitation to the number of calls that can be attempted, so this has to be handled by the softswitch, in conjunction with the IP WAN gateways.

A H.323 gatekeeper can be used to control bandwidth quite effectively, in fact the bandwidth control features of H.323 make that protocol quite attractive in this situation. The voice enabled Cisco routers use the H.323 protocol in conjunction with the gatekeeper to ask permission whenever a call needs to be setup. The actual gatekeeper itself is not Cisco Call Manager (as you might expect) but a Cisco router with gatekeeper software installed. Cisco Call Manager works very well in conjunction with the Cisco gatekeeper by presenting itself to the gatekeeper as another gateway. More information about the Cisco view of call admission control can be found at this link: [VoIP call admission control](http://www.cisco.com/univercd/cc/td/doc/cisintwk/intsolns/voipsol/cac.htm).

If you are looking at a SIP proxy softswitch, then make sure you investigate how your vendor's softswitch instigates CAC. You may find that the CAC scheme is basic or not very scalable.

Step 5: Add IP phones and power them

At last we get to the stage we've all been waiting for. NEW PHONES, and sleek little numbers too mostly. However, there is one little drawback that we didn't need to consider before. Power. IP phones require much more power than their

basic analogue or digital cousins, and this can be a major planning disaster if not handled correctly. Another side effect of this is that our trusty analogue phones powered by a battery backed-up service provider's office were always available for 000 emergency (or 112 or 911, or 111 or). Now we have to consider that we have to be able to provide access to emergency services, so a fully redundant and backed up system is not an option, it is a necessity. Some places continue to keep a single Plain Old Telephony Service (POTS) analogue phone in strategic positions for emergency calls only, rather than going for a fully fail-safe system.

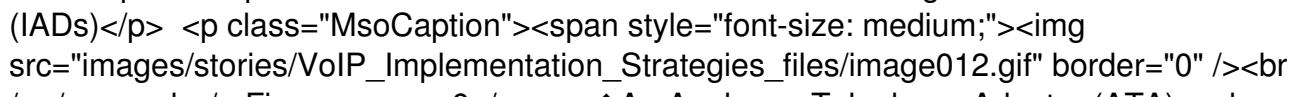
Power options include replacing networking switches with switches that support the IEEE's 802.3af Power over Ethernet (PoE) standard. If this is too expensive, then there are modules that sit in the wiring closet called mid-span PoE devices. Mid span because they inject power into the category 5 Ethernet cabling leading to the phones somewhere between the LAN switch and the phone. Be sure you choose phones that support the standard 802.3af PoE, even if you don't implement it immediately.

Your choice of phone was probably made along with the type of softswitch you chose. If you use Cisco Call Manager, then you're pretty much stuck with Cisco phones, although there are some 3rd party handsets on the market. Most IP phones however support the Session Initiation Protocol (SIP). Even Cisco phones can be purchased with a SIP licence rather than a Skinny (Cisco's proprietary VoIP protocol) licence. Note that Cisco Call Manager version 5.0 supports SIP phones as well, so the 'stuck with Cisco phones' phrase no longer applies in this case.

Adding IP phones with a centralised call agent (softswitch) also adds another dimension to the WAN as well. No longer are the phones at the Regional or Branch offices able to make a phone call if they lose communication with the Central Office. A key component in any design is the fallback scenario should the WAN fail. Cisco uses a technique called [Survivable Remote Site Telephony](http://www.cisco.com/en/US/products/sw/voicesw/ps2169/products_data_sheet09186a00800888ac.html) which works well with Call Manager. If you choose another solution, make sure you ask your reseller how remote site communicate if the WAN is down and you can't reach the softswitch!

Analogue Telephone Adapters (ATAs)

You can add IP functionality to existing analogue phones as well if you want to delay the purchase of the more expensive handsets. Devices known as Analogue Telephone Adapters (ATAs) have an Ethernet port and an FXS (analogue phone) port. Your old phone plugs into the FXS port, and your softswitch controls the ATA as if it were another IP phone. Many ATAs have PSTN bypass ports as well, so that if the WAN goes down, you can still call out via the PSTN. Similarly, if you've kept your PSTN line, calls coming in will still ring the analogue phone. Larger versions with multiple analogue FXS ports are also available, but not as readily as you may think. These devices are sometimes called Integrated Access Devices (IADs).

 Figure 6 An Analogue Telephone Adapter (ATA) makes a good choice for a small office.

The Analogue Telephone

Adapter (ATA) makes a good choice for a small office or home office, especially with only one or two lines. When an external caller dials the local inward number, the analogue phone rings as normal. To make a call, the user picks up the analogue phone, dials digits which are collected by the ATA and sent off to the softswitch (CCM or SIP Proxy) for analysis, just as if the call was being made from an IP phone. On the most upmarket ATA models, it is possible for the ATA to even be programmed to ring the IP phone as well as the analogue phone. There is usually an **override** code that the analogue phone user can dial if they prefer to call directly to the PSTN, and in the event of a power outage, the analogue phone reverts to a direct connection to the PSTN.

Step 6 Remove all legacy equipment

OK. So now you have a full VoIP network, you don't need all those old PABXs or legacy analogue and digital handsets, so have a fire sale, give them to charity, sell them to employees or somehow get someone to take the junk off your hands. But perhaps not so fast. Before you ditch all of your old equipment, make sure that you have complied with all your local regulations about providing Emergency services. You may wish to keep some of those old phones at least, even if you remove the PABXs.

If you have chosen the Cisco Call Manager path, then you will find an add-on module ensuring your VoIP phones have access to Emergency Services, but this is tailored to a North American environment, and don't come cheaply. And of course it depends on the phones being fully backed up during a power outage. If you have chosen a SIP Proxy or other softswitch solution, your best bet is probably to keep some at least one PSTN line connected to an analogue line just for emergencies. Remember that an analogue phone with an ATA can behave like an IP phone most of the time, but in the case of emergency, it can be used to access a local line directly.

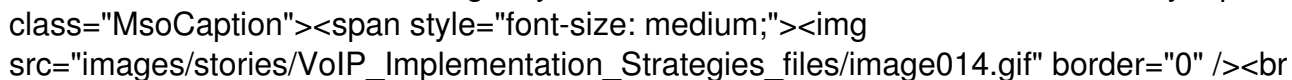


Figure 7 - Completely IP controlled VoIP network.

Your network is now completely controlled by IP softswitches, and all access to the PSTN is now controlled by gateways that also happen to be Cisco routers. But more and more, people are beginning to ask, **Why do we need any access to the PSTN?** Can't our IP service provider take care of our phone calls too? And of course this is where the hottest action is right now. The advent of the **Pure IP** voice service using Voice Service Providers.

Step 7 Remove the PSTN connections & use the services of a VSP

A pure IP network. No need for PABXs, Key systems or Voice enabled routers! You may never reach this stage in the foreseeable future. But then again you might. It depends very much on what is happening behind the scenes in the VSP market. Because it is so easy to become a VSP (if regulations allow) there is fierce competition between a plethora of VSPs right now

[_ftn1](file:///Courses/Nectar/D&I%20Voice%20Enabled%20Network/VoIP%20Implementation%20Strategies/Temp/VoIP%20Implementation%20Strategies.htm#_ftn1)

class="MsoFootnoteReference">[1].Once the big telephony service providers wake up to the fact that they will soon be only providing a backup role in the telephony market, and decide to become serious VSPs themselves there may be some very quick movement. This is already happening in the US, but Australia and NZ are still in startup stage.</p> <p class="MsoCaption">

 Figure 8 ♦ Pure IP network. No need for PABXs, Key systems or Voice enabled routers!</p> <p class="MsoBodyText">Here are a few Aussie VSP websites ♦ not necessarily comprehensive nor even current:</p> <p class="MsoBodyText">Koala Telecom Pty Ltd , GOtalk, VoXaLot , VoipStunt , Talkscape, SIPME, People Telecom, PennyTel, OZtell, Nehos Communications, Mytel, MyNetFone, Koala, Internode - NodePhone♦ , FaktorTel, eVoIP, engin, Broad IP, AstraTEL, SPANtalk. In NZ try Sipserve, Conversant or parrot communications.</p> <p class="MsoBodyText">Links to other providers can be found at http://forums.whirlpool.net.au/index.cfm?a=wiki&tag=VOIP_Providers and voip-info.org at this page.</p> <p class="MsoBodyText">There are also wholesale VoIP providers, such as Soul and <a

<http://www.commsgroup.net.au/>Comms Group Australia, or you can simply join the rest of the SIP based VoIP world for free via <http://www.sipbroker.com/sipbroker/action/login>SIPBroker.

</p> <p class="MsoBodyText">One interesting twist that is emerging from this barrage of VSPs is the new option of going to a VoIP solution without migration from the PSTN at all.Not yet mature enough to be a big corporate business solution, but many small businesses are jumping straight to step 6 ♦ after buying some IP phone equipment of course.</p> <h2>The incumbent PABX vendor model</h2> <p class="MsoBodyText">This model may have many points in its favour, providing you don't have to replace too much antiquated equipment.Not only does it have fewer steps, the risk of failure is somewhat reduced, and you get to deal with your existing vendor, assuming you are happy with their current services.</p> <p class="MsoBodyText">In this model I have migrated the network to Pure VoIP in just 4 steps.As with the previous model, it is assumed that the eventual aim is to completely replace all telephone handsets with IP models in all offices ♦ even if not for some time.In any case, you will at least need to consider the possible bandwidth demands such a migration would have.</p> <p class="MsoBodyText">Also assumed is that you have done your homework particularly with bandwidth and electrical power consumption calculations.</p> <h3>Step1</h3> ♦ Add IP capacity to existing PABXs/Key Systems & upgrade WAN links</h3> <p class="MsoBodyText">No matter what model you follow, the chances are that you will need to upgrade some, if not all, your WAN equipment to cater for the upgraded link speeds you've already calculated, and to implement the kind of QoS (Quality of Service) that you know that your VoIP network demands. In the previous model, we also included voice enabled routers.Stick with the minimum 128Kbs rule for the SOHO situation.The ♦voice enabled router♦ expense is not necessary in this model, (so long as you can get the QoS you require) although some vendors may be confused about this.</p> <p class="MsoCaption">

 Figure 9 ♦ The big expense comes with the PABX and Key System upgrades</p> <p class="MsoBodyText">The big expense instead comes with the PABX and Key System upgrades.This expense can be the death-knell of this approach, especially if you find your existing equipment has to be upgraded.Branches that have old Key Systems are very likely to have to be replaced with small PABXs.If you have to upgrade your routers too, then you get a double whammy.If this is the case, then forget this model ♦ this model assumes that the incumbent PABX vendor is able to ♦add♦ IP to your existing equipment.If not, jump ahead to The IP softPABX (Asterisk) model.</p> <p class="MsoBodyText">Note however, that as yet we have not touched the PSTN connections.The main strength of Incumbent PABX vendor migration model is that we don't need to do anything with PSTN connections at all.</p> <p class="MsoBodyText">The new VoIP enabled PABX is also going to be your Call Control Agent ♦ or softPABX, so we are ready immediately to start adding IP phones.</p> <h4>Call Admission Control (CAC)</h4> <p class="MsoBodyText">Don't forget to investigate how WAN access is going to be controlled by your PABX vendor.If your vendor's PABX supports H.323 gatekeeper and H.323 gateway functionality, then control will probably be

good. If your vendor is suggesting a proprietary or SIP based solution, then make sure you are satisfied with the fail-over functionality of the system.

Step 2

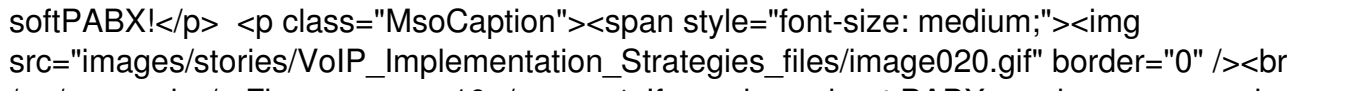
◆ Add IP phones
◆ and power them

Remember that IP phones require much more power than their basic analogue or digital cousins, and this can be a major planning disaster if not handled correctly. Another side effect of this is that our trusty analogue phones powered by a battery backed-up service provider's office were always available for 000 emergency (or 112 or 911, or 111 or). Now we have to consider that we have to be able to provide access to emergency services, so a fully redundant and backed up system is not an option, it is a necessity. Some places continue to keep a single Plain Old Telephony Service (POTS) analogue phone in strategic positions for emergency calls only, rather than going for a fully fail-safe system.

Power options include replacing networking switches with switches that support the IEEE's 802.3af Power over Ethernet (PoE) standard. If this is too expensive, then there are modules that sit in the wiring closet called mid-span PoE devices. Mid span because they inject power into the category 5 Ethernet cabling leading to the phones somewhere between the LAN switch and the phone. Be sure you choose phones that support the standard 802.3af PoE, even if you don't implement it immediately.

Your PABX vendor probably determined your choice of phone. Most vendors will steer you towards a proprietary solution, but some have both proprietary and standards (usually SIP) based solutions. The SIP versions may not have all the features of a proprietary version, so you will have to weigh up the pros and cons of having a standards based solution verses a more fully featured version.

Your vendor may also suggest that money can be saved by centralising call control. After all, you don't need the physical lines in that branch office PABX any more, just logical connections that can easily be handled at the central softPABX. However, adding IP phones with a centralised call agent (softPABX) also adds another dimension to the WAN as well. No longer are the phones at the Regional or Branch offices able to make a phone call if they loose communication with the Central Office. A key component in any design is the fallback scenario should the WAN fail. Make sure you ask your reseller how remote site communicate if the WAN is down and you can't reach the softPABX!

 Figure 10 ◆ If your incumbent PABX vendor can upgrade existing equipment, you can be up using IP handsets with a minimum of fuss.

Depending on your vendor's solution, you may be able to add IP functionality to existing analogue phones as well if you want to delay the purchase of the more expensive handsets. This will usually only work for standards based systems such as SIP. The function of Analogue Telephone Adapters (ATAs) was discussed in detail in the previous section under the heading Analogue Telephone Adapters (ATAs).

The Analogue Telephone Adapter (ATA) makes a good choice for a small office or home office, especially with only one or two lines. On the most upmarket ATA models, it is possible for the ATA to even be programmed to ring the IP phone as well as the analogue phone.

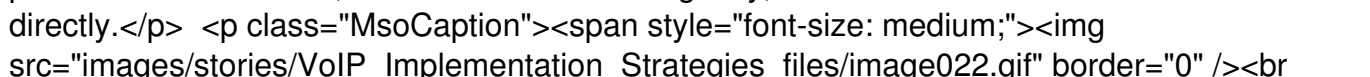
Step 3

◆ Remove all

legacy equipment

Now you have a complete VoIP based network network.


You can gradually remove digital and analogue phones and replace them with IP versions. You will find that there are quite a few digital cards in you PABXs that you don't need any longer either., but before you ditch all of your old equipment, make sure that you have complied will all your local regulations about providing Emergency services. You may wish to keep some of those old phones at least. Remember that an analogue phone with an ATA can behave like an IP phone most of the time, but in the case of emergency, it can be used to access a local line directly.


Figure 11: Completely IP controlled VoIP network with upgraded PABXs.

Your network is now completely controlled by IP softPABXs, and all access to the PSTN is still controlled the way it always has. But more and more, people are beginning to ask, Why do we need any access to the PSTN? Can't our IP service provider take care of our phone calls too? And of course this is where the hottest action is right now. The advent of the Pure IP voice service using Voice Service Providers.

Step 4: Remove the PSTN connections & use the services of a VSP

Apart from the diagram, the final result is very similar to the previous model. What you need to decide is what is the best way to get here? Which vendor can lead me to this point in the way that best suits my company? If you haven't already, read the discussion for Step 7: Remove the PSTN connections & use the services of a VSP in The Softswitch (Cisco Call Manager) model.


Figure 12: pure IP network evolved from your incumbent PABX vendor.

The IP softPABX (Asterisk) model

If you discover that your existing PABX equipment is not easily upgradeable, or you simply want to explore other PABX vendors, or even the open source Asterisk model, then your story will be something like this:

As with the previous models, it is assumed that the eventual aim is to completely replace all telephone handsets with IP models in all offices even if not for some time, and of course you have done your homework particularly with bandwidth and electrical power consumption calculations.

Step 1: Add PABXs & upgrade WAN links

This model shares the concept that new voice gateways (either new vendor PABXs or Asterisk PABXs) will be needed, just like in The Softswitch (Cisco Call Manager) model. The difference of course is that the gateways are not Cisco routers, but IP softPABXs. Like both previous models, the chances are that you will need to upgrade some, if not all, your WAN equipment to cater for the upgraded link speeds you've already calculated, and to implement the kind of QoS (Quality of Service) that you know that your VoIP network demands. The 128Kbs minimum rule still applies, so ADSL is often a good option for SOHO installations.



 Figure 13 ♦ Start by adding the new hardware and program it

Start by installing your upgraded routers and new PABX (or Asterisk) hardware. Make sure that you have programmed and tested your new system before cutting over. When you are ready, you can start moving trunks much the same as in the Softswitch Model.

Inter-Asterisk eXchange (IAX)

IAX is the Inter-Asterisk eXchange protocol used by Asterisk. It can be used as a transport for virtually any type of data, but gets its strength by being able to transport voice much more efficiently than the normal Realtime Transport Protocol (RTP) normally used to carry voice.

The current version of IAX, IAX2, uses a single UDP data stream, usually on port 4569, to communicate between endpoints, both for signalling and voice data. The voice traffic is transmitted in-band, making IAX2 easier to firewall and more likely to work behind network address translation. This is in contrast to SIP, which uses an out-of-band RTP stream to deliver information.

IAX2 supports trunking, multiplexing channels over a single link. When trunking, data from multiple calls are merged into a single set of packets, meaning that one IP datagram can deliver information for more than one call, reducing the effective IP overhead without creating additional latency. This is a big advantage for VoIP users, where IP headers are large percentage of the bandwidth usage. (Source: <http://en.wikipedia.org/wiki/IAX>)

IAX is not exclusive to Asterisk. It has become very popular for trunk links because of its bandwidth efficiency, and several other PABX vendors are now offering IAX2 support.

Step 2 - Add PSTN connections to the IP softPABX

Buy additional circuits and connect them to the PSTN for testing purposes. This means that for some time you will need to have dual circuits while you test the new ones. When you do your homework on the Erlang calculations, you may find that you need fewer PSTN trunks, because in the new model, most of your inter-office traffic will traverse the WAN. After all, this is probably what we are trying to achieve.

 Figure 14 ♦ add extra circuits to your IP softPABX or Asterisk server and test them thoroughly.

When you are sure that the systems are working well, you can migrate the PABX connections from the PSTN to the IP softPABX.

Call Admission Control (CAC)

Don't forget to investigate how WAN access is going to be controlled by your IP softPABX vendor. If your vendor's PABX supports H.323 gatekeeper and H.323 gateway functionality, then control will probably be good. If your vendor is suggesting a proprietary or SIP based solution, then make sure you are satisfied with the fail-over functionality of the system.

If you are using Asterisk, your choices are limited. Asterisk has extensions that allow it to function as an H.323 gateway, which in turn could be controlled by a gatekeeper. At present, Asterisk has no Gatekeeper functionality. Some third party developers may be able to offer CAC on Asterisk, but these are probably not open source (free) implementations.

Step 3

Connect legacy PABX to the new IP PBX

This is where you no longer need the old PABX for any purpose other than to keep connectivity with legacy phones. Since all users in this model still access all numbers via the PABX and all PSTN connectivity is via your new PABX/Asterisk server will probably need as much capacity between the old PABX and the new PABX/Asterisk server as you previously had between the PABX and the PSTN. This may also require you to re-program the old PABX as well, which makes it a particularly vulnerable time.

Since all trunks are now controlled by IP based servers, you can now take advantage of cheaper inter-office and least cost routing for your calls. For many businesses, this is where the major savings are to be gained, and you haven't begun the expensive IP phone replacement. You don't need to have VoIP phones just yet, but you are in a good position to begin IP phone deployment as new handsets are needed or have to be replaced.

 Figure 15 - The old PABX system now serves no purpose other than to keep connectivity with legacy phones

Step 4 Add IP phones and power them

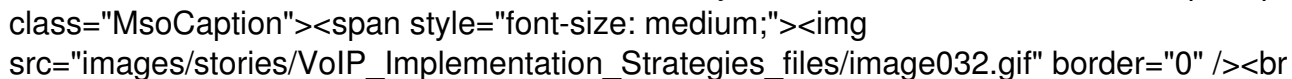
Remember that IP phones require much more power than their basic analogue or digital cousins, and this can be a major planning disaster if not handled correctly. Another side effect of this is that our trusty analogue phones powered by a battery backed-up service provider's office were always available for 000 emergency (or 112 or 911, or 111 or 112). Now we have to consider that we have to be able to provide access to emergency services, so a fully redundant and backed up system is not an option, it is a necessity. Some places continue to keep a single Plain Old Telephony Service (POTS) analogue phone in strategic positions for emergency calls only, rather than going for a fully fail-safe system.

Power options include replacing networking switches with switches that support the IEEE's 802.3af Power over Ethernet (PoE) standard. If this is too expensive, then there are modules that sit in the wiring closet called mid-span PoE devices. Mid span because they inject power into the category 5 Ethernet cabling leading to the phones somewhere between the LAN switch and the phone. Be sure you choose phones that support the standard 802.3af PoE, even if you don't implement it immediately.

Your PABX vendor probably determined your choice of phone. Most vendors will steer you towards a proprietary solution, but some have both proprietary and standards (usually SIP) based solutions. The SIP versions may not have all the features of a proprietary version, so you will have to weigh up the pros and cons of having a standards based solution verses a more fully featured version. If you chose an Asterisk server solution, you would be well advised to stick to SIP based phones, and buy them from a single vendor to keep configuration consistent.

You may also consider that money can be saved by centralising call control. After all, you don't need the physical lines in that branch office PABX any more, just logical connections that can easily be handled at the central softPABX. However, adding IP phones with a centralised call agent (softPABX) also adds another dimension to the WAN as well. No longer are the phones at the Regional or Branch offices able to make a phone call if they loose communication with the Central Office. A key component in any design is the

fallback scenario should the WAN fail. Make sure you ask your reseller how remote site communicate if the WAN is down and you can't reach the softPABX!

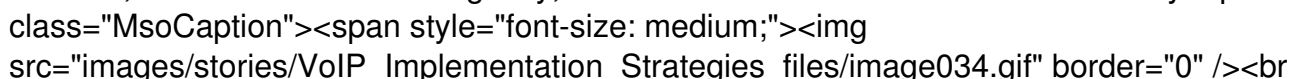


Depending on your vendor's solution, you may be able to add IP functionality to existing analogue phones as well if you want to delay the purchase of the more expensive handsets. This will usually only work for standards based systems such as SIP. The function of Analogue Telephone Adapters (ATAs) was discussed in detail in the previous section under the heading Analogue Telephone Adapters (ATAs).

The Analogue Telephone Adapter (ATA) makes a good choice for a small office or home office, especially with only one or two lines. On the most upmarket ATA models, it is possible for the ATA to even be programmed to ring the IP phone as well as the analogue phone.

[Step 5](#) Remove all legacy equipment

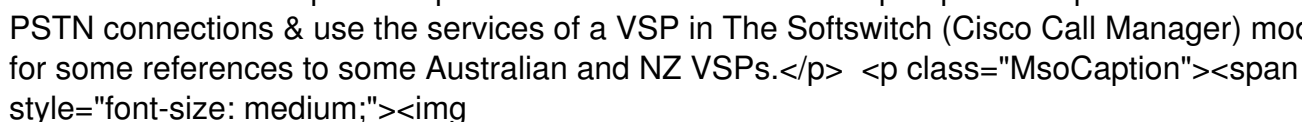
Now you have a complete VoIP based network. You can gradually remove digital and analogue phones and replace them with IP versions, but before you ditch all of your old equipment, make sure that you have complied with all your local regulations about providing Emergency services. You may wish to keep some of those old phones at least. Remember that an analogue phone with an ATA can behave like an IP phone most of the time, but in the case of emergency, it can be used to access a local line directly.

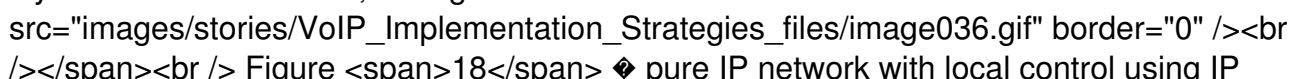


Your network is now completely controlled by IP softPABXs. But more and more, people are beginning to ask, Why do we need any access to the PSTN? Can't our IP service provider take care of our phone calls too? And of course this is where the hottest action is right now. The advent of the Pure IP voice service using Voice Service Providers.

[Step 6](#) Remove the PSTN connections & use the services of a VSP

Not surprisingly, like the models before this, we end up at pretty much the same endpoint. A pure IP network. No need for PABXs, Key systems or Voice enabled routers! You may never reach this stage in the foreseeable future. But then again you might. It depends very much on what is happening behind the scenes in the VSP market. See the discussion for [Step 7](#) Remove the PSTN connections & use the services of a VSP in The Softswitch (Cisco Call Manager) model for some references to some Australian and NZ VSPs.





[Conclusion](#)

One size does not fit all, but please look at these models and see if they

partially fit your scenario. Remember, I pitched these scenarios so that each method had a point (just before adding IP phones) where the VoIP functionality was limited to the WAN, with the intention that that may well be as far as you progress. You may have no intention of using the WAN just yet, and want to start by deploying IP phones for internal local office calls initially, with all external calls via the PABX. These models don't suit that scenario, but you may get some ideas and some point to contemplate by reading the article.

If your intention is to migrate to a VoIP network, starting with VoIP in the WAN, then using The incumbent PABX vendor model has a lot of merit, **providing your existing equipment is easily upgradeable.** If you want to keep the migration as cheap as possible, using an open source system such as [Asterisk](http://asterisk.org/) is tempting, especially if you are competent and comfortable with Linux and enjoy editing text files using vi. If not, there are multiple (open source) user interfaces that can be used with Asterisk, the most advanced being [TrixB](http://www.trixbox.org/), which can turn a spare PC into a running Asterisk system in under one hour.

However, if you want to have a top of the line system, money is no object and you need to choose a trusted vendor, then Cisco CallManager has many benefits. However, most PABX vendors also have some pretty high availability, highly redundant and highly efficient options too.

To get the answer as to which scenario is right for you, your need to ask yourself some questions about how important each of the following items are, then ask your vendor how they can implement each:

[Ten questions](#)

- How are WAN failures handled? Will the remote sites still be able to access the PSTN?
- How is codec selection handled? Do local (within office) calls get a high-quality codec, whereas inter-office office calls get a low bandwidth codec?
- What standards are supported? Is the system using standards based protocols or proprietary protocols that lock you into a single vendor?
- What is the budget?
-

normal; -x-system-font: none;"> What is the timeframe?</p> <p class="MsoBodyText" style="margin-left: 70.75pt; text-indent: -28.2pt;">6. If IP phones are to be deployed, how is power supplied? It is a standards based (802.3af) solution?</p> <p class="MsoBodyText" style="margin-left: 70.75pt; text-indent: -28.2pt;">7. How is emergency services handled in the new environment, especially in the event of a power failure?</p> <p class="MsoBodyText" style="margin-left: 70.75pt; text-indent: -28.2pt;">8. How is Call Admission Control implemented to ensure that voice quality does not degrade due to lack of bandwidth?</p> <p class="MsoBodyText" style="margin-left: 70.75pt; text-indent: -28.2pt;">9. How is QoS implemented to ensure that no amount of data traffic will interfere with voice traffic?</p> <p class="MsoBodyText" style="margin-left: 70.75pt; text-indent: -28.2pt;">10. What support can your vendor deliver when things go wrong? Do they have experienced local staff, or will they be relying on remote assistance?</p> <p class="MsoNormal">◆</p> </div> <div>
 <hr width="33%" size="1" /> <div id="ftn1"> <p class="MsoFootnoteText">[1] July 2006</p> </div> </div> <p>◆</p> <h4>
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