Star2Star Blended Architecture

What Makes it Different? What Makes it Better?

OVERVIEW

In this paper, we discuss:

- What's wrong with the existing telephone network?
- Three ways to switch to VoIP
- What's Different about Star2Star?
- The problem with existing VoIP solutions
- How Star2Star's Blended Architecture works

Over the past few years, the telephone industry has undergone a degree of change that was unthinkable just ten years before. Changes in technology, in the marketplace, and in the overall business climate created a "perfect storm" condition that has seen millions of customers switch their home and business telephone service from the Public Switched Telephone Network (PSTN) to Voice over Internet Protocol, or VoIP.

So much has happened so quickly that customers are confused. They have more choices than ever, and all of the available VoIP solutions appear to do exactly the same thing. But there are huge disparities, both technical and financial, between the available VoIP solutions – especially for business users.

We at Star2Star Communications believe that we have the best solution for businesses of all sizes. In this white paper, we'll explain how our system is different from and better than the other VoIP solutions, and how we're re-inventing the idea of the business phone company for the Internet age.

Why VoIP? Why Now?

VoIP offers a compelling alternative to traditional PSTN telephone lines, especially for businesses with a large number of phone lines or multiple business locations. But VoIP can also save money for smaller businesses with just a few lines. There are many reasons to switch to VoIP, but a few key reasons are:

- VoIP lines typically cost about half as much as a conventional phone line. The savings in monthly line charges can often pay for a VoIP system in a year or two – even sooner for organizations with many phone lines.
- With Star2Star, VoIP lines are virtual, not physical. If a business has dozens, hundreds, or even thousands of locations, they can buy a pool of VoIP lines from Star2Star for all locations to share. This is especially effective for retail outlets, restaurant chains, and other businesses with hundreds or thousands of business locations.
- VoIP makes it easy to unite satellite offices, mobile users, and work-at-home employees into a single system with a single dialing plan. This improves communications within the organization and improves service for their customers.

Even if businesses aren't ready to make the switch to VoIP right now, they'll have to make the switch sooner or later. In a December 2009 filing with the Federal Communications Commission, AT&T said: "*the business model for legacy phone services is in a death spiral.*" The paper went on to lay the groundwork for the eventual phase-out of the PSTN in favor of Internet-based communications. The company once known simply as "The Phone Company" is essentially admitting that VoIP is a better solution.

A Better Way to do VoIP

Once an organization has made the decision to switch to VoIP, they'll have to decide how to make the switch. There are several ways to accomplish the switch. The best path to choose depends on the organization's existing phone system and the amount of time, money, and effort they can commit to making the switch.

In 2006, Star2Star's engineers looked at the then-current VoIP landscape and saw an opportunity. None of the existing approaches to VoIP were "bet your business" reliable. Their solution was to take a completely different approach to VoIP, using a blend of on-site hardware and off-site hosted application servers in a secure, highly reliable data center.

This blended architecture approach to VoIP requires a different business model than competitive systems. Instead of selling PBX systems, IP phones, or VoIP service, Star2Star sells all three, offered together as a unified, end-to-end system. The entire system is reliable, self-monitoring, extremely flexible, and easy to use – while still providing buyers with the advanced features and cost savings they expect from a VoIP system.

The Star2Star system includes all of the features that businesses have come to expect from a high-end PBX system, including unlimited voice mail, auto-attendants, automated call distribution, conference calling, and ring groups – all with no additional monthly fees. The Star2Star blended architecture is especially well suited for businesses with multiple locations, or with many off-campus users.

What's Different About Star2Star?

VoIP is all about standards. So how can one VoIP system be very different from the others? The Star2Star system is built using industry-standard components and protocols, and is fully standards compliant. But Star2Star uses a new type of system architecture that provides users with an end-to-end, fully managed and monitored system.

This unique business model means that Star2Star customers deal with one provider, receive one monthly bill, and have one number to call for service. They didn't create a new business model just to be different. They did it because it is the only way to guarantee customers the same type of seamless, "it just works" experience that they've come to expect.

The StarBox on-premise PBX, the IP telephone sets, and our VoIP telephone service are very tightly integrated with one another. Star2Star maintains controls over and continuously monitors all elements of each customer's system, and the StarWatch monitoring system uses automated sonic testing to spot small problems before they turn into large problems. The StarBox PBX is small, extremely reliable, energy efficient, and has no fans or moving parts.

Star2Star technicians inspect and pre-configure each and every phone and PBX before it leaves the warehouse. They also run a multi-day connection assessment test on each client site before the installation begins. If the connection isn't good enough to provide a reliable VoIP connection, they'll work with the ISP or LEC to correct the situation before allowing the installation to proceed.

What's Wrong with Existing VoIP Solutions?

There are literally hundreds of VoIP solutions available in today's marketplace. Many are simply bundles of off-the-shelf solutions, and others are renamed or repackaged versions of other products. Since all of these solutions appear to do essentially the same thing – connect voice calls over the Internet instead of the PSTN – it is increasingly difficult for buyers to come to grips with the complexity of today's VoIP offerings.

There are three basic classes of VoIP offerings on the market today:

1. SIP trunking for existing PBX systems
2. Hosted IP PBX systems
3. On-premise IP PBX systems

Any of these approaches will get a company connected with VoIP, but they may not be the best approach.

SIP Trunking

PBX systems represent a significant investment, both in terms of capital cost and in user training. Many companies are perfectly happy with their existing PBX systems but would like to enjoy the cost savings of VoIP.

PBX manufacturers have seen VoIP coming for several years now, and most of them offer an option to connect VoIP lines (called SIP Trunks) to their legacy PBX systems. To switch to a SIP trunk solution, customers must disconnect their PSTN service and order SIP trunks from an Internet phone carrier. In most cases, existing phone numbers can be ported over to the new lines.

This seems simple enough, and it has some obvious advantages. First, the system works like it always did. Extension numbers and phone numbers remain the same, and employees already know how to use the system. This solution provides the savings of VoIP without the capital outlay and retraining costs.

But there are several downsides to this approach:

- If the PBX is more than a few years old, it may not make economic sense to upgrade the system. Adding new extensions to an outdated PBX requires making an investment in an essentially obsolete system.

- Voice mail is often a weak point of many existing phone systems, and switching to VoIP won't improve the voice mail situation.

- Many older PBX systems offer limited conference calling features if they offer them at all, and switching to SIP trunking won't change that. To make up for this shortcoming, many companies use a pay-as-you-go conferencing service that adds to their communications costs.

- The call distribution features of older PBX systems are limited, inflexible, and often difficult to program. In fact, they are downright primitive when compared to any modern VoIP PBX.

- A technician usually must perform extension moves and changes, making them costly and time-consuming.

- SIP trunks often do not work well behind some Internet firewalls. NAT routers also pose a problem for SIP traffic. It is often possible to install a single SIP trunk or phone behind a NAT router, but multiple SIP devices may require additional equipment to correctly pass traffic to and from the Internet.

- Many existing PBX systems are a piecemeal solution, with a PBX from one vendor, an ACD system from another and a voice mail product from a third. Put them all together, and they spell "support nightmare."

- Older PBX systems were designed and built with the assumption that workers stay at their desks all day. They can't easily accommodate work-at-home users, mobile phones, or computer-based softphones.

Because they were designed for a circuit-switched world, the feature set of those systems is oriented towards PSTN lines. Desktop set digital displays are usually minimal or lacking entirely, as are the convenience keys that are a standard feature on most VoIP phones. Common operations (transfer, conference, etc.) are often performed using feature codes entered on the keypad.

Many older PBX systems were designed in the days when power consumption and heat output weren't major design considerations, so they often use large amounts of power and they convert much of that power into heat, which must be ventilated. Over the life of the system, the power and heat savings will pay for a significant portion of the cost of a new, more efficient PBX.

Hosted PBX Solutions

Hosted PBX offerings replace an on-premise PBX system with a virtual, hosted PBX. On the surface, this seems to be the easiest solution of all: Remove the outdated PBX system and phones, and put a shiny new SIP phone on every desk, using a Hosted PBX service.

While hosted PBX systems offer much of the functionality of a real, physical PBX, they fall short in several key areas:

- **Bandwidth Allocation & Prioritization**

  While individual VoIP calls don't use a great deal of Internet bandwidth, multiple simultaneous VoIP conversations can have a significant negative impact on the Internet connection. Conversely, heavy data traffic on the Internet connection can tie up bandwidth resources just when they're needed for voice traffic. This can cause delays, stutter, and periods of silence on VoIP calls.

  Traffic shaping techniques can solve these problems by regulating the amount of bandwidth assigned to voice and data traffic on a real-time basis. However, most hosted PBX products can't do this because traffic shaping requires physical access to the Internet connection. Hosted PBX products aren't physically located at the same location as the phones, so they can't alter the flow of traffic on the Internet connection.

- **More Expensive Than They Appear**

  Many hosted PBX vendors charge a monthly license fee for each phone, in addition to the monthly line charges for each SIP trunk. Some hosted PBX providers charge a monthly fee for automatic call distribution, music on hold, and other features that are standard on the Star2Star system. This can make a seemingly inexpensive system much more expensive over a long term.

A SIP PRIMER

At first glance, VoIP and SIP appear to be one and the same. But SIP is actually one of several VoIP protocols, although it is the most widely used. (The H.323 protocol preceded SIP but was not widely adopted.)

Like many Internet protocols, SIP uses a request/response transaction model. The actual SIP commands are human-readable and look very much like the HTTP commands used by web servers.

SIP is only part of the VoIP picture. SIP defines the methodology to create, modify, and terminate voice and/or video connections over the Internet, but it does not handle the actual transmission of the voice stream.

SIP uses the Real-time Transport Protocol (RTP) to actually deliver the VoIP datastream across the Internet.

RTP was designed to deal with the vagaries of Internet connections, so it is designed to handle out-of-sequence and missing data packets. It also includes a *jitter buffer* that compensates for differences in the propagation delay of sequential packets.

In addition to SIP and RTP, VoIP calls typically use a standardized data compression algorithm, commonly called a codec. VoIP codecs efficiently reduce the amount of bandwidth needed for a call without introducing significant delay into the conversation.

- **Sub-optimal Call Routing**

  About two-thirds of the telephone calls in a typical office are intra-office, or intercom calls. These are calls placed from one extension in the building to another. On many hosted PBX systems, intercom calls must travel from the office to the hosted PBX and back again. For example, an intercom call between two phones in the same building in Miami might actually travel all the way to Virginia and back. This ties up trunk resources and Internet bandwidth and reduces call quality.

- **No Support for Legacy Equipment**

  Many companies find that after switching to VoIP, they still need an analog connection for legacy analog equipment such as overhead paging systems, remote door openers, entry phones, and other analog equipment.

- **No Monitoring**

  Hosted PBX systems lack any type of centralized monitoring. When a problem develops with a hosted PBX, it is usually up to the users to detect and report the outage to their service provider.

- **NAT Issues**

  Many hosted PBX products have the same NAT routing issues mentioned in the SIP trunks section. NAT traversal problems can result in missed or dropped calls, one-way audio, and excessive delay or echo.

- **Support Issues**

  Finally, hosted PBX solutions are often a multi-vendor affair, with one company providing the phones, and another providing the virtual PBX and trunks. When something goes wrong, troubleshooting can be a challenge.

Premise-based IP PBX Systems

Premise-based IP PBX systems typically offer higher performance than hosted solutions. Because they are located on the customer's premises, IP PBX systems can use QoS and other traffic shaping techniques to maintain a balance between VoIP and Internet data traffic. They can also make intelligent routing decisions so that in-building intercom calls stay in the building, without traveling over the Internet.

But many IP PBX systems suffer from the same or similar problems as their non-IP counterparts:

- **Local (and often limited) Voicemail and IVR Storage**

  Most IP PBX systems store voicemail and Interactive Voice Response (IVR) recordings on the PBX itself. This means that there is a finite amount of storage space on the PBX, so administrators must keep a vigilant eye on users' voicemail usage. It also means that if the power, PBX, or Internet connection goes down, users will not be able to retrieve their voice mail. Callers will not hear IVR prompts, nor will they be able to leave a message.

- **No Intelligent Multi-site Call Routing**

  The majority of IP PBX systems handle call routing the same way as their PSTN predecessors; they hand the call off to the network, and let the network worry about it. The approach works fine for outbound calls, but it results in less-than-optimal call routing for companies with multiple office locations.

- **No Monitoring**

  Like their hosted counterparts, most IP PBX systems lack centralized monitoring features. Without continual monitoring, problems can go undetected and unreported for hours or even days.

- **Limited Failover Options**

  IP PBX systems typically represent a single point of failure. If the PBX or Internet connection goes down, the system can't handle incoming or outgoing calls. Most competitive IP PBX systems have limited failover provisions; others sell failover as an expensive add-on.

- **Reduced Reliability**

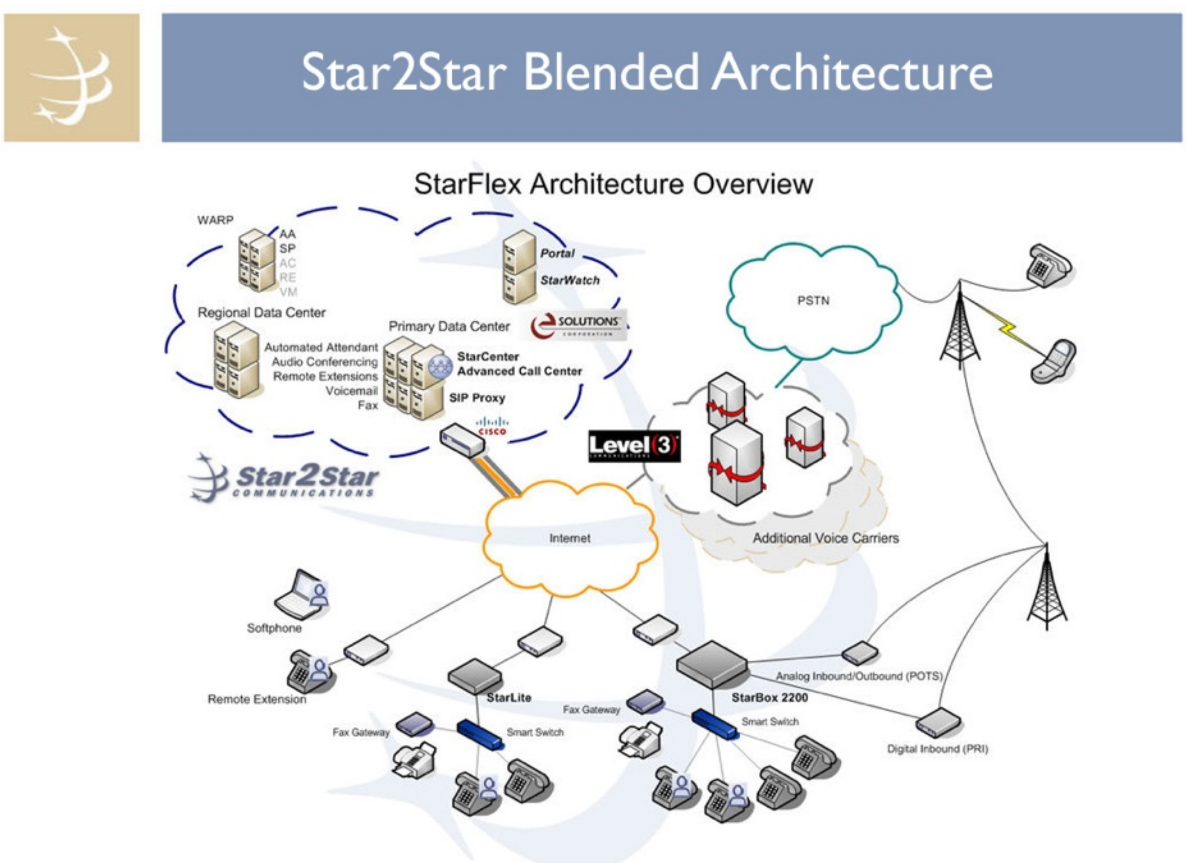
  Some IP PBX products are built using commodity PC hardware. While this helps to hold costs down, most PCs are not designed for mission-critical 24/7/365 operation. Even high-reliability server-class PCs use fans and hard drives with a limited lifespan.

The StarFlex Architecture

Star2Star's unique blended architecture combines an on-premise, no-moving-parts PBX system with a suite of services (hosted at their own highly-reliable, redundant data centers) to create a complete, feature-rich communications system. Despite a large number of features, the Star2Star system is extremely easy to use. System management tasks can be performed from any PC with a web browser, anywhere.

The on-premise portion of the Star2Star system is the StarBox, a self-contained digital PBX. There are several StarBox models for different-sized offices. The StarBox is compact and energy efficient. It produces very little heat, so it doesn't need a noisy, unreliable fan for ventilation. The StarBox operating software runs from flash memory, so there's no spinning hard drive to wear out or fail unexpectedly.

On the inside, the StarBox is based on an open-source PBX system. The StarBox itself runs on AstLinux, a compact version of Linux tailored specifically for PBX systems. The StarBox also runs our own proprietary traffic shaping, configuration, and monitoring software. The StarBox can connect to a single WAN connection, or it can connect to multiple WANs for added reliability and failover protection.



PSTN VS. VOIP

In the PSTN world, a "line" is a circuit that is delivered to a customer, either over a discrete pair of wires (on a POTS circuit) or as part of a bundle of circuits delivered over a T1 or other multiplexed carrier. Either way, those circuits connect a customer's PBX system to a telephone company switch. If a customer will potentially need to handle 100 calls at any given time, then they'll need 100 circuits.

Star2Star phone lines aren't "lines" at all, although we call them that because that's what our customers are used to. Our VoIP "lines" are *virtual* circuits between a client's PBX and our call center. With Star2Star, clients order as many lines as they expect to need on a peak basis - for all of their business locations, combined. We can even add extra circuits "on the fly" to handle seasonal or unexpected traffic increases.

For example, a large national retail chain recently converted to Star2Star. They have nearly 9,000 stores, and each store had a phone line. After the switch, they have 3,000 Star2Star lines. We ported over all of their 9,000 phone numbers, so each store still has a unique local number for incoming calls.

The StarBox connects to the Internet and to the office LAN, and it manages all of the VoIP traffic between the local LAN and the Internet. This allows the StarBox to add Quality of Service (QoS) information to Internet traffic so that time-sensitive voice traffic gets priority over data traffic. The StarBox accomplishes this without requiring any modifications to, or reconfiguration of the existing LAN or Internet connection. The end result is clearer, more intelligible calls with no distortion or delay.

Most StarBox systems include a backup power supply and a managed Ethernet switch with Power over Ethernet (PoE.) This allows individual telephones to receive power over the same Ethernet cable that is used to connect the phone to the LAN. PoE eliminates the need for an AC power outlet and telephone power transformer at each telephone location, which provides a cleaner, more attractive installation.

Star2Star desktop phones include a pass-through Ethernet connection, so customers can use a single cable for a computer and a telephone at the same time. VLAN technology keeps the PC and telephone traffic separate.

Many telephone system vendors offer only a handful of telephone sets. Star2Star offers dozens of models of phones from leading IP phone vendors, including Polycom, snom, and Cisco. Customers can choose from desktop phones, conference room speaker phones, single-line phones, and even cordless phones.

Adapters are available to connect to legacy telephone lines and equipment including overhead paging systems, analog "2500-style" telephone sets, and existing T1 and/or POTS trunks. A POTS trunk can be used as an emergency failover line in the event of an Internet outage.

StarService Internet Telephone Service

The StarBox PBX connects over the Internet to the StarCenter Network Operations Center, or NOC. There are six NOCs located around the country. Each NOC is located in a highly secure facility, with redundant power, triple-redundant Internet connectivity, and strict security measures. Each NOCs is in the same building with several Internet telephone carriers, so calls can be switched directly to those carriers' networks with virtually no additional routing.

The NOCs also contain servers for voice mail, audio conferencing, automated attendant, and other system features. These features are hosted on high-reliability servers with multiple backups. Everything is continually monitored to minimize downtime and maximize call quality.

All calls (except for in-building intercom calls) pass through one of our NOCs. Outbound calls are passed on to the appropriate carrier for delivery. Incoming calls may be directed to a customer's StarBox for delivery to a specific telephone, or they may be redirected to an automated attendant, a call queue, voicemail, or an off-premise phone.

This distributed approach reduces the amount of traffic on each customer's Internet connection. For example, an incoming call entering a call distribution queue uses very little bandwidth until someone actually answers the call. All of the voice prompts, music on hold, and other announcement messages the caller hears are generated at the NOC, not in the StarBox. Similarly, voicemail is recorded and stored on a server in the NOC, not at the customer's site.

Star2Star's unique Wide Area Replication Protocol (WARP) monitors the state of the network and the primary carrier's network, and it uses best-path-routing technology to optimize the path for each call. Because the network is tightly integrated with Level 3 (the primary carrier), they can identify the shortest, most responsive path for both incoming and outgoing calls. A unique Best-path routing technology continually monitors the call quality at each of Level 3's six PSTN gateways. When a Star2Star customer places a call, the call is routed using the path that provides the highest call quality, not just the shortest path or the one with the least latency.

Wherever possible, SIP and RTP media traffic are sent over the best and most appropriate route for each traffic type. Media data is only proxied or transcoded when absolutely necessary; this "hands-off" approach maintains the highest possible call quality.

Intercom calls between multi-office locations (and, in fact, all calls between any two Star2Star customer sites) are routed over the Internet without using any SIP trunk resources.

Star2Star System Features

The combination of the StarFlex architecture and the StarService IP telephone service provides users with a broad set of system features in a system that is highly reliable and expandable. Despite all the technical sophistication, the Star2Star system is extremely easy to configure and use.

Standard Star2Star system features include:

- Unlimited Auto Attendants, call queues, and ring groups
- Best-in-class multiple location operation; call centers and hunt groups can span multiple office locations
- Remote (off-campus) phones and PC/Mac-based softphones operate the same way as local extensions
- System wide conference calling with optional password protection
- Find Me / Follow Me roaming keeps users reachable, even outside of the building
- Dynamic line allocation reduces monthly costs; line bursting adds additional lines on the fly when needed for overflow traffic
- Simple, easy to use interface for end users; easy web-based configuration for system managers

Star2Star Support

Despite the high reliability of the Star2Star system, there are times when customers may need a little help. Star2Star operates a 24-hour customer support center at our headquarters in Florida. Every Star2Star system automatically monitors itself and reports potential problems to our support center. These problems may be as simple as an unplugged phone or as serious as a power surge or lightning strike.

In many cases, our support staff knows about problems before our customers do. If the problem is at our end, they will initiate steps to remedy the problem. If the problem is a bad phone or StarBox, they'll arrange to ship a replacement unit out as soon as possible, often the same day. When a component needs to be replaced, we ship one out overnight express, at no charge to the customer. We even provide a prepaid return carton for the defective item.

Most problems are resolved with a single call. And since Star2Star controls both ends of the connection, we take complete responsibility for the entire phone system.

After the Switch: The Star2Star User Experience

Perhaps the best benefit of a Star2Star telephone system is one that's hard to measure. It's the user experience; the way people interact with the system and with one another on a daily basis.

The Star2Star system has literally hundreds of features to help handle incoming and outgoing calls in the most efficient and effective way possible. Our customers have told us a few of their favorite things about using the Star2Star system:

- **Everyone is reachable again**
  Thanks to our innovative find me / follow me call forwarding, even highly mobile employees are never more than a few rings away. Employees can change their own call routing as their needs change.

- **Employees make better use of their time**
  Incoming caller ID and unlimited voice mail lets workers decide which calls to take and which ones to leave for later.

- **Meetings are a thing of the past**
  Flexible, on-the-fly conference calling lets workers attend meetings without leaving their desks. Mobile and work-at-home users can join in, too.

- **Multi-location businesses work as one**
  Satellite offices, work-at-home workers, and traveling sales staff are all part of the same phone system. All workers can share one incoming number, or key employees can have their own numbers.

- **Multi-city presence**
  Each department, office, or remote office can have its own number, even across multiple area codes. Customers can dial a local number in their own city to connect to a supplier across the country.

- **Reducing the voicemail burden**
  Our voicemail-to-email option delivers voicemail messages to an e-mail, Blackberry or iPhone inbox, so users can listen and respond to important voicemails immediately.

- **Custom call handling**
  Our web-based configuration manager makes it easy to create automated attendants and call distribution groups that guide callers to the proper person without an attendant or operator.

- **Happy Customers**
  Star2Star scored a remarkable 93% on a recent customer satisfaction survey conducted by Dun & Bradstreet. This type of score is rare in any business, but it is unheard of among telecom providers.

About Star2Star:

Founded in 2006, Sarasota, Florida – based Star2Star Communications is re-inventing the phone company for the Internet age. Our Star2Star Internet business telephone system provides users with innovative features to increase productivity while decreasing monthly communications costs. Star2Star products are sold exclusively through a network of installing dealers. We offer before- and after-sales support, number porting assistance, and comprehensive dealer training. Despite a tough economy, we're growing at over 300% per year.

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