White Paper

Revolutionizing Communications for the Internet Age

Star2Star Blended Architecture

OVERVIEW
In this paper, we discuss:

- What's wrong with the existing telephone network?
- Three ways to switch to Unified Communications
- What's Different About Star2Star?
- The problem with existing Voice over IP Solutions
- How Star2Star's Blended Architecture™ Works

Over the past several years, the telecommunications 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 combined to create a "perfect storm" condition that has compelled millions of customers to switch their home and business telephone service from the Public Switched Telephone Network (PSTN) to newer, Internet-based communications technologies.

These technologies, collectively called *Unified Communications*, or UC for short, have made substantial inroads towards replacing the 100-year-old technology that powers the traditional telephone network.

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

Adding to the confusion is the fact that different vendors use different terms to define their own vision of what UC is or should be. However you define it, we believe that we have the best UC solution for businesses of all sizes. In this white paper, we'll explain how our system is different from and better than other UC solutions, and how we're re-inventing the idea of the business phone company for the Internet age.

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

- Voice over IP connections typically cost about half as much as a conventional phone line. The savings in monthly line charges can often pay for a new phone system in a year or two – even sooner for organizations with many phone lines.
- Star2Star phone lines are virtual, not physical. If a business has dozens, hundreds, or even thousands of locations, they can buy a pool of 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.
- Our technology makes it easy to unite satellite offices, mobile users, and work-at-home employees into a single, unified system with a single dialing plan. This improves communications within the organization and improves service for their customers.

Terminology Alert

The terms *UC* and *Voice over Internet Protocol*, or *VoIP* are often used interchangeably.

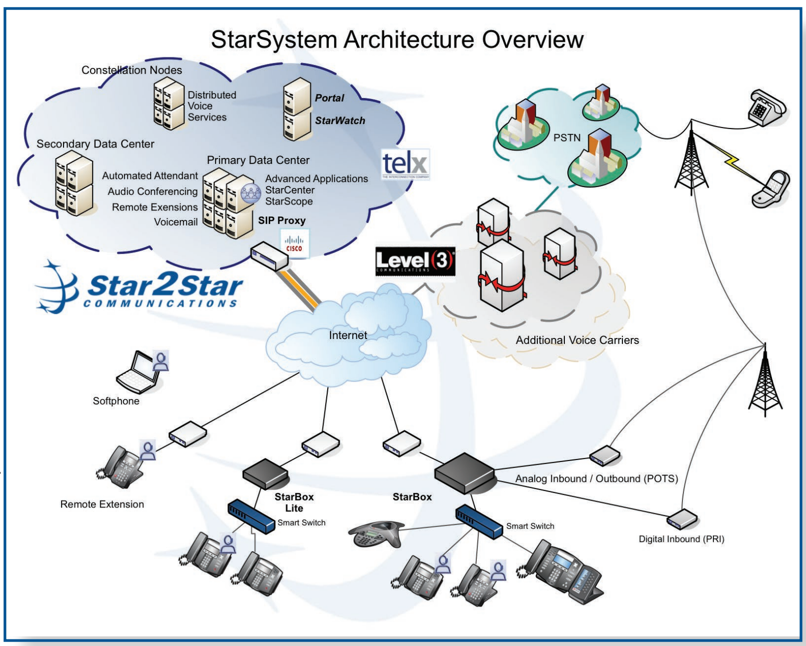
VoIP defines a method for delivering telephone service over the Internet. VoIP is an important and essential component of a UC system, but a true UC system includes messaging, presence monitoring, fax, chat, video conferencing., and other technologies.

Even if businesses aren't ready to make the switch to UC right now, they'll have to make the switch at some point. 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 acknowledging that UC is a better solution.

A Better Way to do UC

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

In 2006, Star2Star's engineers looked at the then-current business telephone 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 the problem, using a blend of on-site hardware and cloud-based application servers in a secure, highly reliable data center.



This Blended Architecture™ approach to VoIP requires a different business model than traditional systems. Instead of selling PBX systems, IP phones, or VoIP service, Star2Star sells all three, designed together and sold together as a unified, end-to-end solution. 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 UC system.

The Star2Star system includes all the features that businesses have come to expect from a high-end business telephone system, including full-featured voice mail, auto-attendants, find-me / follow-me call routing, conference calling, and ring groups. Star2Star's Blended Architecture is especially well suited for businesses with multiple locations, or with many off-campus users.

What's Different About Star2Star?

UC is all about industry standards. So how can one UC system be much different from another? The Star2Star system is built using industry-standard components and protocols, and is fully standards compliant. The difference is that Star2Star's Blended Architecture provides users with an end-to-end, fully managed and monitored system.

This unique design means that Star2Star customers deal with one provider, receive one monthly bill, and have one number to call for service. Customers are guaranteed the same type of seamless, "it just works" experience they've come to expect.

The StarBox™ on-premise appliance, the IP telephone sets, and Star2Star telephone service are very tightly integrated with one another. Star2Star maintains controls over and continuously monitors all elements of each customer's telephone system. The StarWatch monitoring system uses automated sonic testing to spot small problems before they turn into large problems.

The StarBox Internet Communication Appliance is small, extremely reliable, and energy efficient. Star2Star technicians inspect and pre-configure every phone and StarBox before it leaves our order fulfillment center. Our technicians run a multi-day connection assessment test on each client site before installation begins.

If the customer's Internet connection isn't good enough to provide a reliable voice connection, we'll work with the customer's Internet Service Provider or Local Exchange Carrier 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 UC solutions on the market today:

- SIP trunking for existing PBX systems
- Hosted IP PBX systems
- On-premise IP PBX systems

Any of these three approaches will get a company connected with UC, but there is much more to providing a complete business-class solution.

Solution #1: SIP Trunking with an existing PBX

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.

Traditional 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 disconnect their PSTN service and order SIP trunks from an Internet SIP provider. In most cases, existing phone numbers can be ported over to the new lines. SIP trunks can also be connected to legacy systems using an external SIP adapter.

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 while SIP offers customers a relatively simple transition to VoIP, it is not a complete UC solution:

- SIP trunking brings the cost savings of VoIP to an existing telephone system, but it can't provide the complete feature set of an advanced UC system.

- Voice mail is often a weak point of older 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 incoming call routing features of older PBX systems are relatively limited when compared to a modern, full-featured UC system.

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

Many existing telephone systems are a piecemeal solution, with a PBX from one vendor, a call distribution system from another and a voice mail product from a third. Troubleshooting such a system can be challenging.

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 IP phones. Common operations (transfer, conference, etc.) are often performed using feature codes entered on the keypad.

Finally, 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. They convert much of that power into heat, which must be ventilated. Over the life of the system, the power and heat savings can pay for a significant portion of the cost of a new, more efficient system.

Solution #2: Hosted PBX Systems

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 annoying 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.

- **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 the associated Internet bandwidth and reduces call quality.

- **Limited or No Support for Legacy Equipment**
  Many companies find that after switching to a UC solution, they still need an analog connection for legacy equipment such as overhead paging systems, remote door openers, entry phones, and other analog equipment. Many hosted systems don't offer this capability.

- **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 do not work well behind a NAT firewall. NAT traversal problems can result in missed or dropped calls, one-way audio, and excessive delay or echo.

- **No Failover Provisions**
  Hosted PBX products typically have limited or no failover provisions. If any part of the system suffers a catastrophic failure, the entire system can go down.

Solution #3: 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 Quality of Service 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 shortcomings 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.

A SIP Primer

If you spend any amount of time reading about UC and VoIP, you'll come across something called Session Initiation Protocol, or SIP.

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 can 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.

- **Limited or 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.

The StarFlex Blended Architecture

Star2Star's unique Blended Architecture combines an on-premise IP communications appliance with a suite of cloud-based services (hosted at 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 StarBox Family of IP Communication Appliances

The on-premise portion of the Star2Star system is the StarBox, a self-contained digital communication appliance. There are several StarBox models for different-sized offices. All models are compact and energy efficient; the smallest draws only 6 Watts of AC power. The StarBox operating software runs from flash or Solid-State Disk memory, so there's no spinning hard drive to wear out or fail unexpectedly.

On the inside, StarBox runs on AstLinux, a compact, security-hardened 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 utilize multiple WANs for added reliability and failover protection.

The StarBox connects between the Internet and the office LAN, and it manages all of the VoIP traffic between the 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 Star2Star systems include 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.

The use of PoE eliminates the need for an AC power outlet and a power transformer at each telephone location, providing a cleaner, more attractive installation.

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

Many telephone system vendors offer only a handful of telephone sets. Star2Star offers a wide selection of phones from leading IP phone vendors, including industry leaders Polycom 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 (analog) trunks. A POTS trunk can be used as an emergency failover line in the event of an Internet outage.

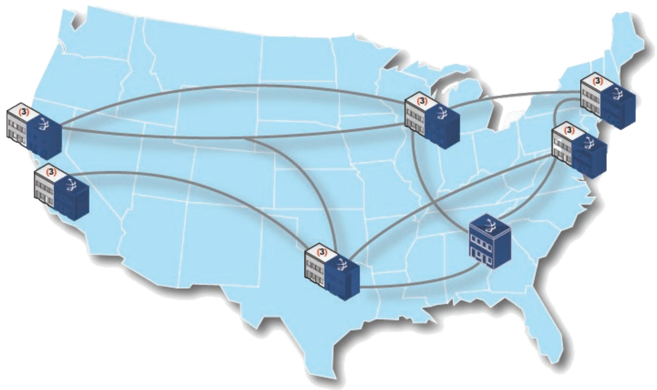
StarService Internet Telephone Service

The StarBox PBX connects over the Internet to Star2Star's Constellation network. As its name implies, Constellation utilizes an array, or constellation, of call routing nodes and data centers, distributed across the USA.

The Constellation technology continually collects data about the performance of each node, as well as the performance of the Internet traffic surrounding each node. It then uses data collected from the nodes to determine the best call routing to use for each call placed on every Star2Star customer system.

Constellation assures users of the best possible call quality by dynamically determining the best route for calls to take as they traverse the Internet. The result is crisp, clear, audio on every call, with none of the annoying echo or delay that affects many other VoIP providers.

The Star2Star data centers are located in highly secure facilities, with redundant power, redundant Internet con-



nectivity, and strict security measures. Each data center is co-located with many Internet telephone carriers, so calls can be switched directly to those carriers' networks with virtually no additional routing.

The data centers contain servers for cloud-based voice mail, audio conferencing, automated attendants, 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 the Constellation network. 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 greatly 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 data center, not in the StarBox. Similarly, voicemail is recorded and stored on a server in the data center, not at the customer's site.

Star2Star's unique Constellation architecture continually monitors the state of the 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), Constellation 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 simply the shortest path or the one with the least latency.

SIP or RTP?

VoIP traffic consists of two components, each handled by a different set of protocols.

SIP is used to connect and disconnect calls. The actual digitized voice data (called the *media*) of a VoIP call is carried by the Real Time Protocol, or RTP for short.

Most VoIP providers send their SIP and RTP over the same path. Star2Star's Constellation technology can utilize different data paths for SIP and RTP, and can even switch the RTP path "on the fly" in order to maintain call quality.

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.

Finally, 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 trunk resources.

Star2Star Software

The Star2Star Application Framework is a powerful, robust application platform. It provides fast, efficient distribution of cloud-based software and data. The framework allows Star2Star to deliver application software - and the application's associated data - to any PC or Mac, anywhere there's an Internet connection.

So, for example, if you sign in to the framework from your office PC, all of your Star2Star applications are instantly available. If you're working on your Mac at home for the day, those same applications and data are waiting for you when you sign in. Wireless LAN at the airport? Check.

There are currently three applications (called Starlets) that run in the Star2Star Application Framework, with more on the way:

- StarScope 2 - Star2Star's exclusive Integrated Communications tool
- StarFax Personal - Full featured, cloud-based fax with your own personal fax number
- Messages (included with StarScope 2) - Point-and-click access to your voicemail, recorded calls, and incoming faxes

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:

- 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, and iPhone 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.

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. Our end-to-end guarantee includes next-day equipment replacement.

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 Star2Star:

- **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 your email 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 in Sarasota, Florida, Star2Star Communications develops and delivers Integrated Communication and Collaboration solutions that connect and enable Productive Business People. Star2Star's award-winning, patent-pending technology overcomes the reliability and quality limitations of other Internet communications technologies. Star2Star Communications systems unify customers' voice, fax, and instant messaging communications into a single, easy-to-use system.

During 2011, Star2Star achieved 100% network uptime and was named to both the Inc. 500 and Forbes Most Promising Companies lists.

Star2Star products are sold through a diversified international network of distributors, master agents, and certified installing dealers. Available across North America, Star2Star Internet communication systems are installed in tens of thousands of businesses and in many large national retail and restaurant chains.

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