Cisco Systems

White Paper

SIP: The Promise Becomes Reality

Introduction

The Internet and mobile communications have had a profound impact on individual and business communications methods. Changes have come rapidly, and according to DataQuest, the amount of traffic on packet networks surpassed that of time-division multiplexing (TDM) networks in the year 2000.

According to Jupiter Media Metrix (2001), the use of instant-messaging applications at work increased 48 percent in 2001—from 2.3 billion to 4.9 billion minutes—making it the most rapidly growing technology in history. Cell phones, pagers, and personal digital assistants (PDAs) are no longer reserved for those with technology curiosity; they are now a mainstream way for coworkers, friends, and family to reach each other. Many people rely on mobile and wireless devices to access essential information quickly and easily. They find that they are better prepared and more effective in their work and personal lives.

In response to these changes, service providers are building next-generation networks that can easily and flexibly accommodate the numerous ways people communicate. They are seeking methods to systematically integrate services and customers in a way that maximizes profitability.

To meet the demands of the changing business environment, to attract new customers, and to add to their portfolio of revenue-generating services, service providers are deploying converged voice-and-data services based on the Session Initiation Protocol (SIP). With its foundation in Internet protocols, SIP provides the ability to integrate traditional voice services with Web-based data services, including self-based provisioning, instant messaging, presence, and mobility services.

Leaders in the communications industry are developing new products and services that rely on SIP, and they are offering attractive new communications services to their customers. Microsoft recently added support for SIP clients in core product offerings—Windows XP and Windows Messenger—a step that will proliferate SIP clients on personal computers worldwide. SIP is gaining momentum in every market, and carriers that have deployed SIP in their networks include interexchange carriers such as Worldcom and Genuity, telephony application service providers and communications service providers such as TalkingNets and TellMe, alternate carriers such as Vonage, Internet telephony service providers such as deltathree, and advanced service providers such as Microsoft—all Cisco customers.

Cisco is enabling the advance of new communications services with a complete SIP-enabled portfolio, including proxy servers, packet voice gateways, call control and signaling, IP phones, and firewalls. These products are available today. Only Cisco is dedicated to providing ubiquitous and seamless protocol interoperability in its packet-telephony solutions. Cisco solutions support a variety of call-control and standard protocols, including H.323, Media Gateway Control Protocol (MGCP), and SIP, which can coexist in the same customer network.

This paper discusses the SIP protocol and its suitability as the foundation for next-generation networks, defines the role of SIP in a multiprotocol network, discusses the evolution of Cisco SIP products and solutions, and reviews a variety of revenue-generating services currently being deployed.

The Origin of the Session Initiation Protocol

In March 1999, the Internet Engineering Task Force (IETF) defined SIP in RFC 2543. The definition was the culmination of years of work in the IETF's MMUSIC Working Group to provide a mechanism to allow voice, video, and data to be integrated over the same network. SIP allows seamless integration with existing TDM networks, and also allows integration with e-mail, the World Wide Web, and next-generation technologies such as instant messaging and 3rd Generation Partnership Project (3GPP) mobile networks.

Since its inception, RFC 2543 has undergone numerous changes and advancements, and work continues to further define and advance SIP. The IETF has expanded to three working groups:

- *SIP-WG*—Defines the underlying SIP protocol and all extensions to this protocol
- *SIPPING-WG*—Defines common application usage of SIP
- *SIP for Instant Messaging and Presence Logical Extensions (SIMPLE)-WG*—Defines how SIP can be used to build instant messaging and presence. (Both Microsoft/MSN and AOL have publicly announced that they will embrace SIMPLE as the standard for their next-generation instant-messaging products. This will provide the technology to allow the two largest instant-messaging communities to interoperate.)

The most notable SIP characteristics are its usage of an Internet-like distributed architecture and its reuse of existing Internet technologies:

- Domain Name System (DNS) and URLs for naming
- HTTP/text for message formatting
- Session Definition Protocol (SDP) for capabilities exchange
- Multipurpose Internet Mail Extensions (MIME) for application integration

With these capabilities, SIP not only integrates with other Internet technologies, it also allows Web developers to quickly integrate applications.

Another foundation characteristic of SIP is that it is both network and application agnostic. Although most deployments of SIP will be over IP networks, it can also be used for non-IP networks such as ATM. In addition, because it uses SDP to communicate the capabilities exchange of the session, it can be used for voice, video, text messaging, or instant messaging and presence.

Aside from focusing on standardizing SIP within the IETF, the telecommunications community has placed interoperability at the top of its priority and requirements lists. To facilitate interoperability among technology vendors' equipment and service-provider networks, the SIPit event (formerly called SIPBakeoff) is held three times a year. Although only six companies attended the first SIPBakeoff in 1999, the most recent SIPit event—the ninth—had over 70 vendors and service providers in attendance. Commitment to interoperability has allowed SIP devices and networks to be rapidly deployed in production networks around the world.

Cisco has shown a commitment to the standardization and interoperability of SIP by not only attending every SIPBakeoff and SIPit event since the program's inception, but by also maintaining a co-chair of the SIPPING-WG and submitting numerous Internet drafts to expand the functionality of RFC 2543. A list of drafts supported by Cisco is included in the reference section.

SIP Components

A SIP-based network is made up of several components:

- *SIP user agent*—Any network endpoint that can originate or terminate a SIP session; this might include a SIP-enabled telephone, a SIP PC client (known as a "softphone"), or a SIP-enabled gateway
- *SIP proxy server*—A call-control device that provides many services such as routing of SIP messages between SIP user agents
- *SIP redirect server*—A call-control device that provides routing information to user agents when requested, giving the user agent an alternate uniform resource identifier (URI) or destination user-agent server (UAS)
- *SIP registrar server*—A device that stores the logical location of user agents within that domain or subdomain; a SIP registrar server stores the location of user agents and dynamically updates its data via REGISTER messages
- *SIP location services*—Additional functionality that can be used by proxy, redirect, and registrar servers to find the identity (with a unique URI) and "logical" location of user agents within the network
- *Back-to-back user agent*—A call-control device that provides routing similar to a proxy server, but allows centralized control of the network call flows; this device allows SIP networks to replicate certain traditional telephony services that require centralized knowledge of device state, such as call park and pickup; this component is always dialog "stateful"
- *SIP-aware network devices*—Devices that have knowledge of the SIP protocol and allow the network to function more efficiently; this type of device might be a firewall or Network Address Translation (NAT) device that can allow SIP traffic to traverse network borders, or a load-balancing switch that allows requests to SIP servers to be more efficiently handled

The Role of SIP in a Multiprotocol Network

Momentum is growing in the packet telephony market. Since packet telephony technology was introduced in 1997, total packet-based international long-distance usage has grown from 8 million minutes to 3.7 billion minutes, and a 2001 Telegeography report estimates that packet telephony usage over international borders will exceed 6.7 billion minutes in 2002.

Today packet telephony traffic traverses networks that rely on a variety of signaling protocols, including H.323, SIP, and MGCP.

Recognizing the need for solutions that can interconnect and interoperate with other networks regardless of the signaling protocol used, Cisco provides solutions that simultaneously support these standard protocols.

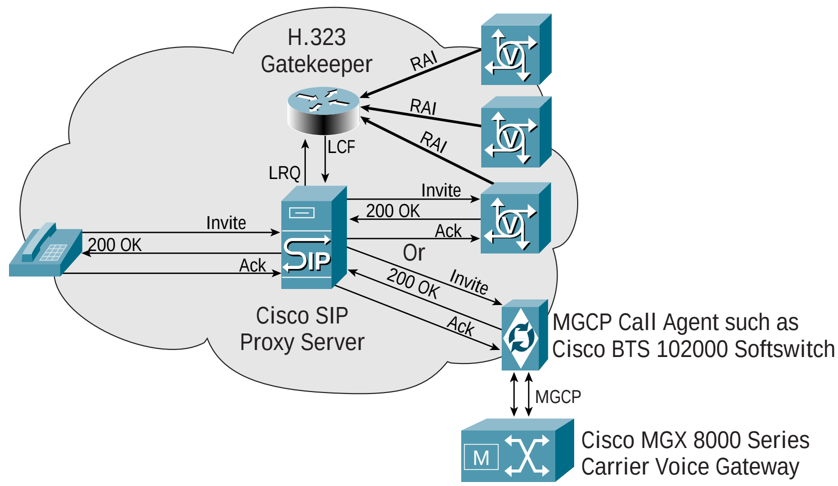
Cisco's multiprotocol strategy has several facets. First, Cisco has implemented support for SIP in its gateway product lines that have existing support for H.323 and MGCP. This extension of Cisco gateway functionality has allowed Cisco to take advantage of the rich features of these gateways, including a broad variety of coder-decoder (codec) support, VoiceXML, sophisticated quality-of-service (QoS) algorithms, and existing Operation, Administration, and Maintenance (OA&M) capabilities.

Second, Cisco offers solutions that enable service providers to deploy services based on different protocols using the same network infrastructure components. For example, Cisco gateways utilizing Cisco IOS® Software can be configured to run H.323, SIP, or both simultaneously. In Cisco networks, Cisco SIP proxy servers and Cisco H.323 gatekeepers communicate effectively to efficiently manage gateway sources in a mixed-protocol environment.

Figure 1 shows an example of how this communication is achieved by using the H.323 protocol resource availability indicator (RAI) mechanism. Cisco SIP proxy servers query H.323 gatekeepers for route status using the H.323 location request (LRQ) mechanism. In this illustration, the SIP protocol is used for internetwork signaling, including connection to a Cisco softswitch, and MGCP is used between a softswitch and network gateways.

Figure 1

Coexistence of H.323, SIP, and MGCP



Cisco strategy will eventually make it possible to build truly converged networks where services built using different protocols can seamlessly interoperate. This goal will be achieved by continuously enhancing the Cisco packet voice architecture to provide protocol interoperability at the signaling layer, and at the same time engineer common solutions at application, OA&M, transport, and security layers.

The Cisco strategy offers three major benefits to service providers. First, service providers can invest in packet-telephony technology based on their opportunity to generate revenue regardless of protocol. Second, this architectural flexibility increases the types of services that can be deployed quickly. Third, this strategy provides the framework for building truly converged networks where services based on H.323, SIP, MGCP, or Public Switched Telephone Network (PSTN) protocols can seamlessly interwork.

Cisco's Broad Array of SIP-Enabled Products

Cisco is committed to helping service-provider and enterprise customers build flexible, robust networks for voice, video, and data, and Cisco has delivered the industry's most complete infrastructure solutions to deliver services based on SIP.

Cisco's SIP-enabled product portfolio encompasses all components of a SIP network infrastructure, from IP phones and access devices to call-control and PSTN interworking. Cisco's first SIP products were deployed with live traffic almost three years ago. All the following products are deployed in live networks spanning a variety of applications and continents.

- *Cisco IP phones*—Cisco IP series phones, including the Cisco IP Phone 7960 and the Cisco IP Phone 7940, support SIP user-agent functionality. These IP phones deliver functionality such as inline-power support and dual-Ethernet ports, and deliver traditional desktop functionality such as call hold, transfer, conferencing, caller ID, call waiting, and a lighted message-waiting indicator.
- *Cisco ATA 186 Analog Telephone Adaptor*—The Cisco ATA 186 supports SIP user-agent functionality. With two foreign-exchange-station (FXS) ports and a single Ethernet port, the Cisco ATA 186 provides a low-cost means to connect analog phones to a SIP network. It also delivers traditional desktop functionality such as call hold, transfer, conferencing, caller ID, and lighted call-waiting and message-waiting indicators.

- *Cisco packet voice gateways*—Cisco 1700 Modular Access Routers, those that are voice capable, Cisco 2600 Series multiservice platforms, Cisco 3600 Series multiservice platforms, Cisco AS5000 universal gateways, and Cisco 7200 Series voice gateways all support SIP user-agent functionality. They provide a means of connecting SIP networks to traditional TDM networks via T1, E1, DS3, channel-associated signaling (CAS), Primary/Basic Rate Interface (PRI/BRI), R2, FXS, foreign exchange office (FXO), or ear and mouth (E&M) interfaces. Cisco packet voice gateways are used to build the largest packet telephony networks in the world.
- *Cisco SIP Proxy server*—The Cisco SIP Proxy Server provides the functionality of a SIP proxy, SIP redirect, SIP registrar, and SIP location services server. The Cisco SIP Proxy Server provides the foundation for call routing within SIP networks; it can interwork with traditional SIP location services such as DNS or ENUM, with feature servers via a SIP redirect message, and with H.323 location services using standard LRQ messages. The Cisco SIP Proxy Server runs on either Solaris or Linux operating systems.
- *Cisco BTS 10200 Softswitch*—The Cisco BTS 10200 provides softswitch functionality to Class 4 and Class 5 networks, and provides SIP-to-Signaling System 7 (SS7) gateway functionality for American National Standards Institute (ANSI) standardized networks. The Cisco BTS 10200 supports SIP user-agent functionality in conjunction with a Cisco packet voice media gateway such as a Cisco AS5000 Universal Gateway or Cisco MGX® 8000 Series Voice Gateway.
- *Cisco PGW 2200 PSTN Gateway*—The Cisco PGW 2200 provides softswitch functionality for Class 4 networks, as well as Internet offload and SIP-to-SS7 gateway functionality for international networks. The Cisco PGW 2200 supports ISDN User Part (ISUP) certification in over 130 countries. The Cisco PGW 2200 supports SIP user-agent functionality in conjunction with a Cisco packet voice media gateway such as a Cisco AS5000 Universal Gateway or Cisco MGX 8000 Series Voice Gateway.
- *Cisco Secure PIX® Firewall*—The Cisco Secure PIX Firewall is a SIP-aware networking device that provides firewall and NAT functionality. Because it is SIP aware, it is able to dynamically allow SIP signaling to traverse network and addressing boundaries without compromising overall network security. A Cisco Secure PIX Firewall functioning in this capacity is called an application layer gateway (ALG).

Networks and Services Deployed Using Cisco SIP Solutions

Many services are now being deployed using Cisco SIP-enabled products, and major carriers have deployed this technology. Some of networks and services currently being deployed include:

- *Interexchange carriers (IXCs)*—WorldCom IP Communications is a breakthrough service, changing the way business is done in the digital world. WorldCom's IP Communications service combines voice and data applications on one network, allowing customers to transition efficiently from a traditional voice platform to IP-based communications.

  With IP Communications, customers can deploy SIP phones without PBX equipment, with call routing and feature control occurring in the WorldCom SIP-based network. Users can maintain familiar voice features, such as call forwarding, calling privileges and call screening. SIP also enables vital new features, such as voicemail that allows users to receive recorded messages via the Web or email. To ensure a smooth migration path, IP Communications can also interoperate with a customer's existing PBX and the Public Switch Telephone Network (PSTN). Customers also receive Web accounts to control feature administration and view call detail online
- *Tier 1 Internet service providers (ISPs)*—Genuity has deployed one of the world's largest packet voice networks, with a worldwide fiber backbone spanning more than 17,500 miles of routes in the United States alone and carrying more than 100 million minutes of use (MOUs) per month. The Genuity network presently covers more than 80 percent of the U.S. metropolitan market and can support 80,000 simultaneous voice calls. Genuity serves two markets: other service providers and enterprise customers.

  Although most VoIP networks use the H.323 protocol, Genuity planned to support SIP as well, because of its increasing popularity. To build its groundbreaking network, Genuity deployed a Cisco Global Long Distance solution including more than 700 Cisco AS5300 Voice Gateways and more than 100 Cisco 3640 and 3662 H.323 gatekeepers.

  Support for multiple protocols in its Cisco Powered Network allows Genuity to attract and interconnect with more customers, supporting whichever control and signaling protocol the customer prefers to use. Integrated support for SIP has enabled Genuity to offer termination services for SIP-based application service providers and to support enhanced voice services such as unified messaging, presence management, IP Centrex, and IP call centers.

- *Application service providers and communication services providers (ASPs and CSPs)*—Taking advantage of the strengths of SIP as an Internet-based protocol, ASPs are deploying a variety of enhanced origination and termination services. Services that fall into this category include conferencing, hosted call centers, voice portals, and debit card applications.

  Among the first to offer managed business services using SIP technology is TalkingNets, a North Carolina-based telephony ASP. By building its network infrastructure with SIP-enabled products from Cisco and its partners, TalkingNets entered the market for voice-based information services at a fraction of the cost of incumbent local exchange carriers (ILECs) and alternate carriers.

  Support for multiple standard protocols, including H.323, SIP, and MGCP in products provided by Cisco has enabled TalkingNets to differentiate itself through enhanced voice services such as unified messaging, the ability to forward voice-mail messages to e-mail as '.wav' files, and presence management. This solution is based on Cisco SIP-enabled products in combination with the Broadsoft Broadworks platform.

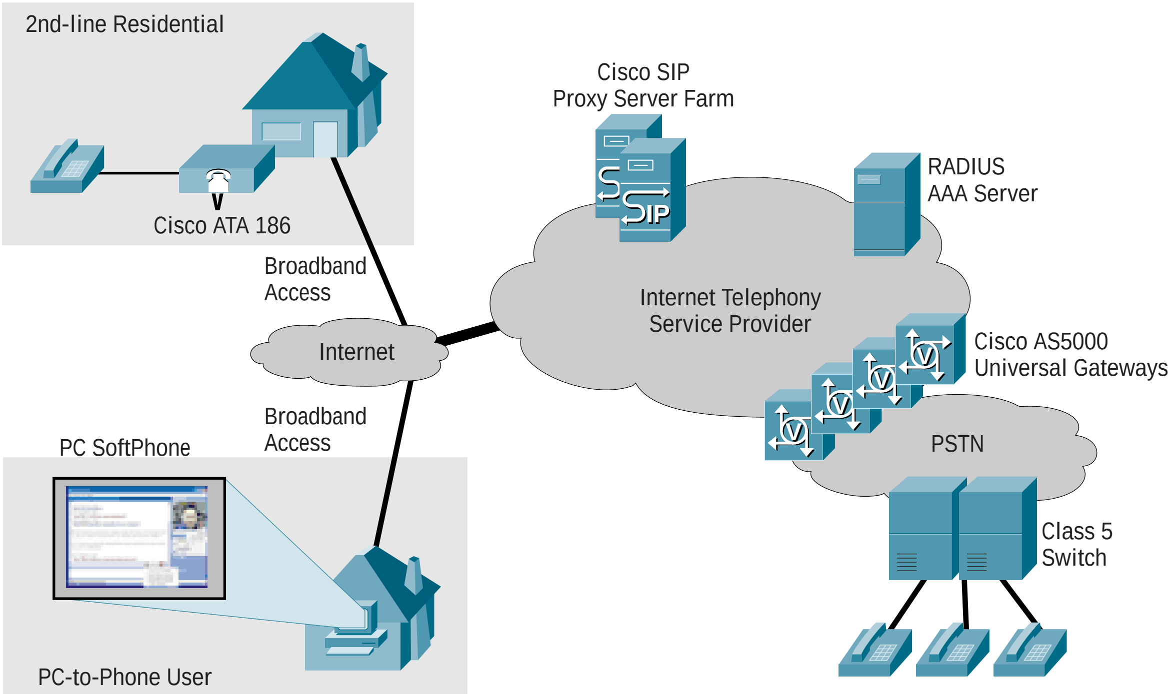
  Another example of an ASP that has begun to deploy SIP-based services is Tellme Networks, Inc. Tellme recently launched "SIP Tellme Studio," the first VoIP-accessible VoiceXML developer resource, available now on the public Internet at sip:8005559655@sip.studio.tellme.com. Tellme Studio is a free VoiceXML development community that allows any developer to build, test, and run VoiceXML applications.

  Tellme is using SIP-enabled gateways from Cisco to allow outbound PSTN calls associated with this service. According to Don Jackson, vice president of advanced telephony at Tellme, "We have a significant investment in interconnection to TDM/PSTN networks. VoIP is clearly the future of telephony and at Tellme we are already running applications on these next-generation networks. We believe that SIP, combined with VoiceXML, is an incredibly robust solution for the enterprise. VoiceXML as the foundation language for voice applications can benefit from the addition of SIP and VoIP to yield powerful new capabilities that allow us to incorporate VoiceXML into even more networks and devices. Cisco clearly recognizes this potential and by integrating SIP into many of their networking products make them great solutions for Tellme to work with."
- *Alternate carriers*—Alternate carriers are actively deploying both managed business and residential services based on Cisco SIP-enabled products.

  Vonage, a leading provider of carrier-grade VoIP communications solutions, is working with Cisco to seize the market opportunity created by the proliferation of broadband access infrastructure. Vonage went live with its residential second-line services, dubbed Vonage Direct, on January 15, 2002. This unique VoIP service offering turns a broadband DSL or cable connection into a second phone line with numerous value-added capabilities. "The confluence of universal access, the growth of broadband connections, and the advent of VoIP has created a large and compelling market," says Carlos Bhola, president of the Edison, New Jersey-based firm.
- *Internet telephony service providers (ITSPs)*—Probably the most well-known SIP-based applications provided by ITSPs today are the variety of PC-to-phone offerings based on Microsoft's Windows Messenger, a SIP-based communication client first released with Microsoft Windows XP. Figure 2 shows a high-level architecture for this solution.

Figure 2

A Typical Cisco Packet-Telephony Network Architecture Supporting Microsoft Windows Messenger



Select service providers that have received certification from Microsoft to offer voice-termination services are providing solutions based on Cisco SIP infrastructure solutions.

One of these providers is deltathree. Founded in 1996, deltathree provides Internet telephony services and infrastructure for service providers, small businesses, and consumers worldwide. Service-provider customers private-label deltathree's voice-over-IP (VoIP) services, which include PC-to-phone, phone-to-phone, and broadband phone solutions. When deltathree updated its infrastructure with a Cisco SIP-based solution and became one of the first carriers to adopt SIP, it slashed application development time by half and reduced operational costs 30 to 40 percent.

deltathree has recently announced its intent to standardize on SIP as its protocol of choice. Consistent with this direction, deltathree is expanding its service offering to include prepaid calling and wholesale call transport services. These services will also be based on Cisco SIP-enabled products.

Conclusion

Taking advantage of its foundation in Internet protocols, SIP enhances the ability to create services that integrate telephony with Web-based services such as instant messaging and presence. Though SIP has its roots in Internet technology, it facilitates seamless interoperability with traditional telephony networks and helps packet-based networks achieve the scaling, reliability, and voice quality of the global switched telephone network.

The telecommunications industry has matured since SIP was introduced three years ago, and now the promise of SIP—a lightweight, Internet-friendly protocol that provides the ability to create converged services that integrate traditional voice-based services with Internet-based services—has become a reality. Major carriers in every geographical market have deployed SIP technology, proving that SIP is mature and world ready.

As part of Cisco's multiprotocol voice strategy, Cisco has developed a broad offering of SIP-enabled products and solutions. With the broadest portfolio of packet telephony products and solutions in the market today, Cisco enables service providers to cost-effectively take advantage of the strength of standards-based protocols to reliably scale networks and manage them effectively. The protocol flexibility inherent in Cisco solutions gives service providers the ability to offer a broader service portfolio, including traditional voice services and new services such as IP Centrex, unified messaging, hosted call center, and second-line residential services.

References

1. http://www.ietf.org/internet-drafts/draft-ietf-sip-refer-02.txt
2. http://www.ietf.org/internet-drafts/draft-ietf-sip-replaces-00.txt
3. http://www.ietf.org/internet-drafts/draft-levy-sip-diversion-03.txt
4. http://www.ietf.org/internet-drafts/draft-biggs-replaces-01.txt
5. http://www.ietf.org/internet-drafts/draft-ietf-sip-refer-02.txt
6. http://www.ietf.org/internet-drafts/draft-mahy-sip-message-waiting-02.txt
7. http://www.ietf.org/internet-drafts/draft-ietf-sip-cc-transfer-05.txt
8. http://www.ietf.org/internet-drafts/draft-rosenberg-sip-3pcc-03.txt
9. http://www.ietf.org/internet-drafts/draft-ietf-sip-100rel-04.txt
10. http://www.ietf.org/internet-drafts/draft-ietf-sip-mib-03.txt
11. http://www.ietf.org/internet-drafts/draft-sterman-aaa-sip-00.txt
12. http://www.ietf.org/internet-drafts/draft-ietf-sip-privacy-03.txt
13. http://www.ietf.org/internet-drafts/draft-ietf-sip-manyfolks-resources-03.txt
14. http://www.ietf.org/internet-drafts/draft-mahy-sip-signaled-digits-o1.txt
15. http://www.ietf.org/internet-drafts/draft-mule-sip-t38callflows-02.txt
16. http://www.ietf.org/internet-drafts/draft-ietf-sip-session-timer-08.txt
17. http://www.ietf.org/internet-drafts/draft-ietf-sip-call-flows-05.txt.
18. http://www.ietf.org/internet-drafts/draft-polk-sip-resource-02.txt

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