

White Paper

The SIP Contact Center
Leveraging the SIP Architecture to build
IP contact center solutions

Version 1
October 2007



SIP Contact Center

White Paper

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Introduction

The legacy call center has been a staple for enterprises that service customers over the telephone. Whether servicing a major airline or a small local business – providing prompt access to helpful agents is an important part of every business and a constant target for improving performance and efficiency. As a result, the legacy call center has gone through a metamorphosis, emerging as the IP Contact Center which replaces the PBX and separate IVR and ACD systems and merges email and instant messaging into a new architecture that integrates these functions, leveraging Voice over IP technologies. With IP Contact Centers, full featured contact centers can be deployed with agents virtually anywhere in the world where there is IP connectivity. Whether the goal is to reduce costs in the existing call center, leverage inexpensive overseas labor or add Work At Home Agents (WAHA), IP Contact Centers provide tremendous flexibility to adapt to changing markets and labor resources.

The Legacy Contact Center

The legacy contact center was built around TDM PBX equipment, adding a number of contact center application functional blocks as separate point solutions, each of which had to integrate into the PBX. Integration was complex and relied on proprietary CTI links or proprietary extensions to the signaling protocols on the T1/E1 links.

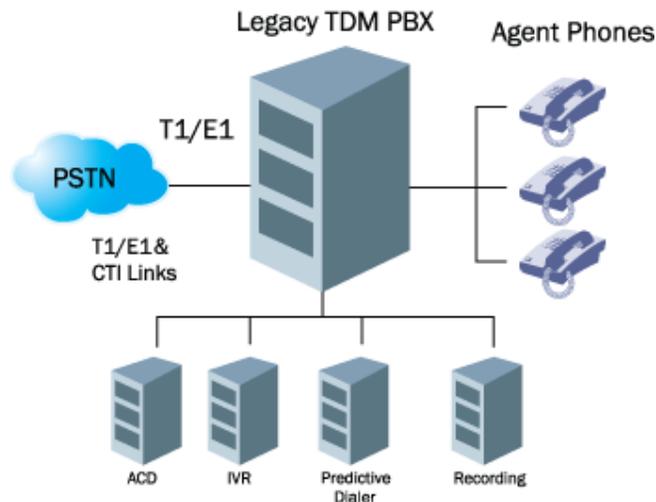


Figure 1: Legacy Contact Center Architecture

Each of the application functional blocks that implemented Automated Call Distribution (ACD), Interactive Voice Response (IVR) and other application features were almost universally built on the Legacy CTI Architecture. This architecture used proprietary APIs, device drivers and board technologies that were ground-breaking in their time, but now are widely recognized as obsolete and suffering from a number of serious limitations.

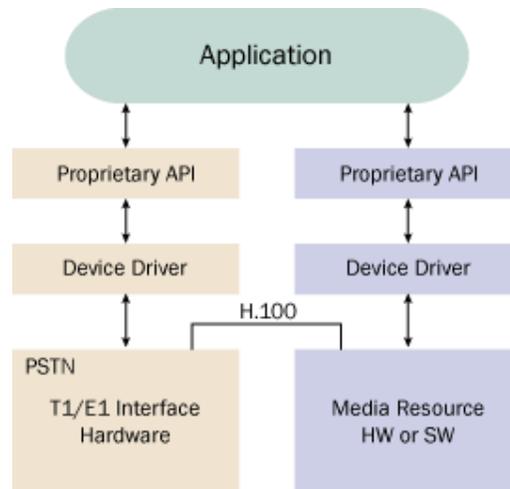


Figure 2: Legacy CTI Architecture

Proprietary APIs - As the hardware and software components in this architecture originated from a number of different vendors, the APIs that control them are very different from one another and very complex. To work in this environment, developers require very specialized knowledge.

Proprietary Device Drivers - The device drivers provided by the vendors are tightly tied to the operating system. The net of this is that the enabling technology vendors dictated which operating systems the developer could use.

Limited Packaging Options – Because the CTI architecture depends on the API to communicate with boards or software via device drivers, all the resources must reside in the same server. This limits features and drives up costs.

Scaling – Once an installation grows to the point where the resources required outgrow the hardware platform, adding more boards and/or servers is usually complex and expensive.

Pace of Change – Due to the complexity, modifications to applications based on APIs is extremely difficult and requires significant effort to completely test and debug development issues. This can delay new features and capabilities, leaving room for competitors to capture market share.

The SIP Architecture

The SIP Architecture approach resolves these issues by using SIP to standardize the interfaces between functional blocks, whether they are DSP boards, separate servers or software modules. The architecture separates the application execution, the connectivity and the media processing requirements into a number of separate functional blocks as shown below:

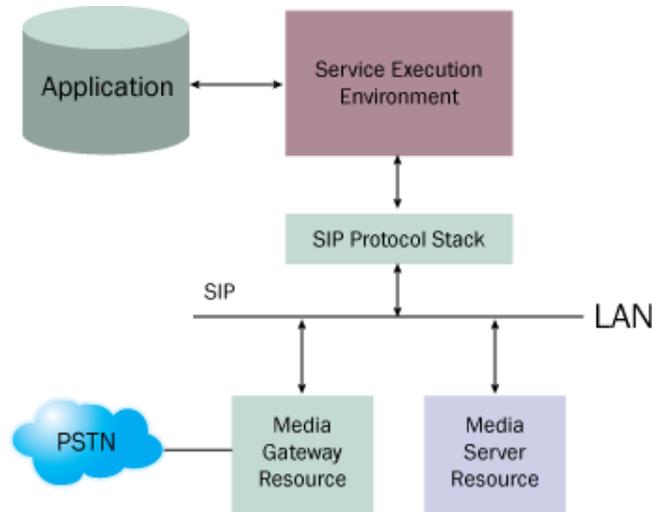


Figure 3: SIP Architecture for PSTN

The SIP architecture has a number of advantages that are important to both the developers and customer solution designers:

Industry Standard Interfaces – by leveraging open protocol standard of SIP (RFC 3261), NetAnn (RFC 4240) and MSCML (RFC 4722), developers have access to a wide range of products, each offering different capabilities from various vendors – all sharing common interfaces.

Operating System and Platform Independence – by using SIP as a control protocol and not using proprietary device drivers, virtually any operating system and any hardware platform can be used to create applications.

Broad Packaging Options – applications based on the SIP architecture can leverage a wide range of physical packaging options. From PCI-based commercial servers, to AdvancedTCA and other blade server form factors, to pre-packaged appliances.

Scaling - the SIP architecture solves a number of scaling issues. Application servers can manage multiple media gateways and/or multiple media servers – allowing the integrator to scale up or down over a very broad range.

Agility and Time to Market – because the SIP architecture is more modular, application modifications can be implemented and deployed more efficiently, helping developers add features and remain competitive in a fast-paced marketplace.

All of these advantages add up to a very compelling argument for adopting the SIP architecture in contact center solutions.

Adopting SIP in the Contact Center - The Driving Forces

A number of driving factors in both business and technology have made the SIP architecture viable for use in a wide range of contact center solutions:

Distributed Workforce - With increasing pressure on costs of staffing a quality contact center, managers have had to think beyond the typical sea of cubicles and adopt new strategies for finding, housing and retaining good agents. Costs of office space, energy and commuting expenses have all shifted the balance point for both employer and employee. Work At Home Agents (WAHA), remote work centers, outsourcing and mobility have created the need for new technology that can accommodate dynamic new work environments.

Protocol Evolution - SIP and the companion media control protocols **NetAnn** and **MSCML** have evolved over the last few years to finally encompass sufficient capabilities needed to implement contact center applications. Prior to these standard protocols, developers had no choice but to use complex proprietary APIs to create the contact center application elements. The new capabilities of these protocols have made complete SIP solutions possible.

Best-of-Breed Standardized Components - The business of building telecommunications equipment has forever changed. Instead of building the application and all of the underlying components themselves, the equipment manufacturers have recognized the value of staying focused on their application feature set, leveraging standard off-the-shelf enabling technology building blocks. SIP has been a key technology for the equipment manufactures to accomplish the move to standardized components and leveraging best-of-breed enabling technologies from a range of suppliers.

The SIP Contact Center

When applied to the IP Contact Center, the architecture takes the form as shown below, creating the SIP Contact Center:

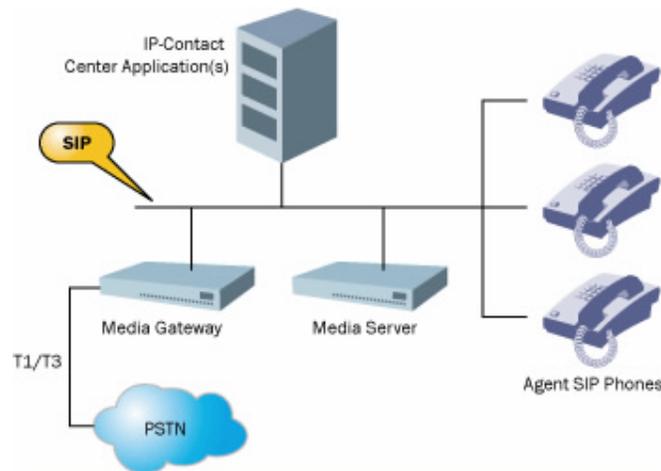


Figure 4: The SIP Contact Center Architecture

Architecture Overview

Following the SIP architecture as a guideline, the SIP Contact Center includes a number of elements:

Application Server Element

The Application Server provides the core logic and management of the contact center solution. Instead of completely separate systems, the functions of the Application Server are usually divided into a number of separate software modules, all working together:

- SIP Protocol Stack – provides connectivity between the Application Server and other elements within the contact center.
- SIP Server – responsible for call switching, transfers and routing. Based on the implementation, the SIP Server function may be either completely integrated into the application or an adjunct software module from other commercial or open-source vendors.
- ACD – inbound call distribution and routing
- IVR - DTMF and possible voice driven and customer-developed interactive voice response applications
- Outbound Dialer – calling customers/prospects and detecting both live and answering machine answer conditions.
- Messaging – voicemail, email and instant messaging

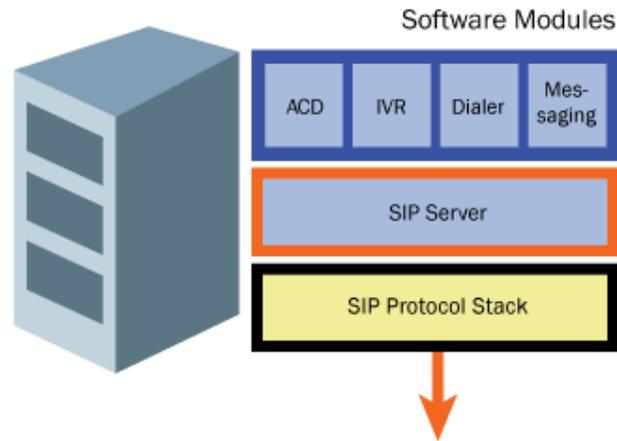


Figure 5: Software Modules within Application Server

Based on the size of the contact center, one or more physical servers may be used to build the solution. The servers may execute just one of the software modules or combined together in a clustered server model - spreading the workload over two or more servers.

Trunking Media Gateway Element

Positioned between the Public Switched Telephone Network (PSTN) trunk lines and the IP Contact Center, trunking gateways provide connectivity to the outside world. All telephone calls that go into and out of the contact center via the PSTN pass through a trunking gateway.

Trunking media gateways have a number of duties, including:

- PSTN Signaling
- Detecting ring voltage
- Converting and compressing the voice into packets
- Echo cancellation
- Tone detection and generation
- Call Progress and Answering Machine detection
- IP control protocols
- Network performance measurement
- And more...

Agent-side Media Gateways

In some applications, a significant investment exists in the analog telephone equipment installed on the contact center agent's desks. As a result, many customers want to continue to use these phones with the new IP Contact Center. Retaining the analog phones for the agents has the added benefit of expedited installation and avoiding the cost of new IP-based phones and their wiring. In this situation, one or more analog FXS media gateways would be used as part of the solution.

Media Server Element

The media server plays a key role in handling all of the media processing needed for the contact center:

- Playing or streaming of announcements and prompts for IVR applications
- Music on Hold streaming
- Conferencing for consultations with supervisors or "subject matter experts"
- Whisper coaching
- DTMF detection and generation
- Call progress tone detection
- Answering Machine Detection

- Training
- Monitoring
- Recording / Logging
- Transcoding

Sample Inbound Call Flow

“A picture is worth a thousand words”

As the saying is true, so seeing the interaction between the different elements in typical applications in a pictorial form should be very useful in building an understanding of how SIP impacts the IP Contact Center. In the series of high-level call flow diagrams below, a real-world series of typical contact center activities is played out as follows:

A customer calls into the contact center and is initially prompted for an account number. The caller presses “0” and the application then connects the caller with an agent. Because of the complexity of the customer’s question, the agent activates a consultation with a supervisor and records the call for future training.

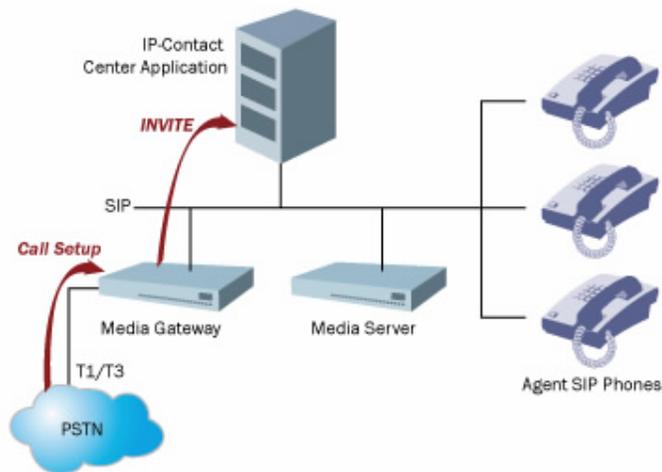


Figure 6: Inbound Call Flow - Step #1

A customer calls into the contact center – the media gateway converts the ISDN Call Setup message from the PSTN to a SIP INVITE and forwards the message to the SIP Server within the application.

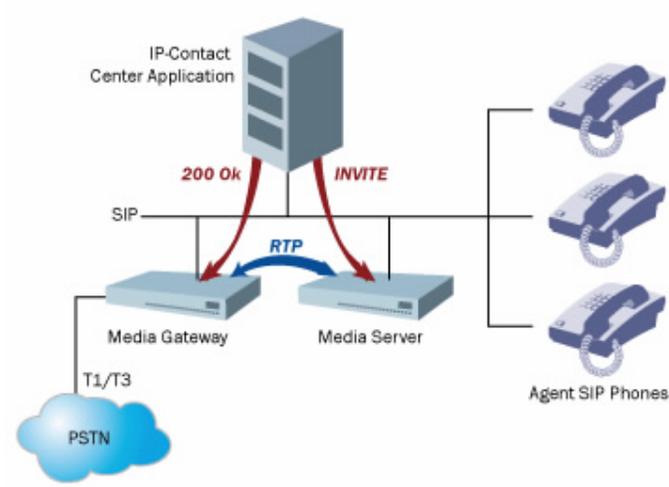


Figure 7: Inbound Call Flow - Step #2

Based on the existing traffic ,load and other factors, the application directs the call to one of the media server resources within the installation using an INVITE to the media server and 200 Okay to the media gateway. This establishes an RTP stream between the caller and the media server resource.

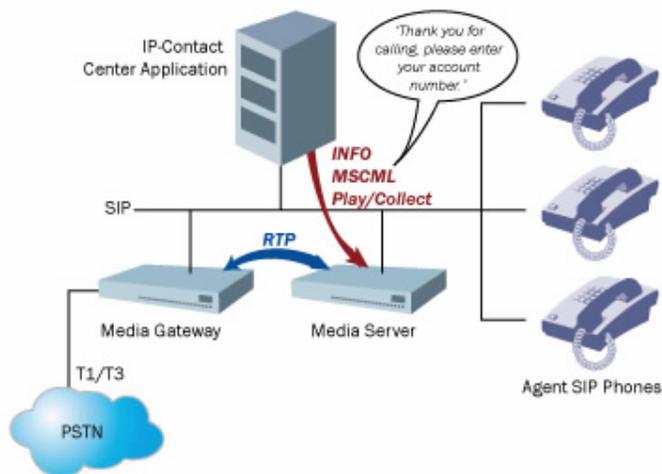


Figure 8: Inbound Call Flow - Step #3

The application then uses the media server and an MSCML Play/Collect command to play an announcement to the caller, asking for their account number.

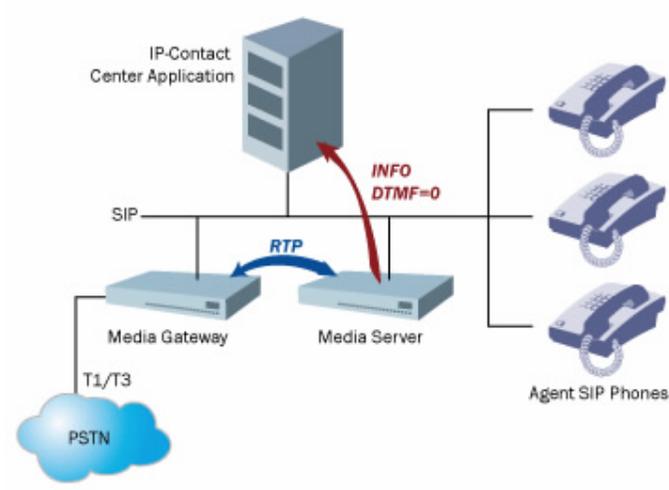


Figure 9: Inbound Call Flow - Step #4

However, the caller has other plans and wants to talk to an agent right away and presses “0”. The media server detects the DTMF, terminates the collection and forwards an event back to the application.

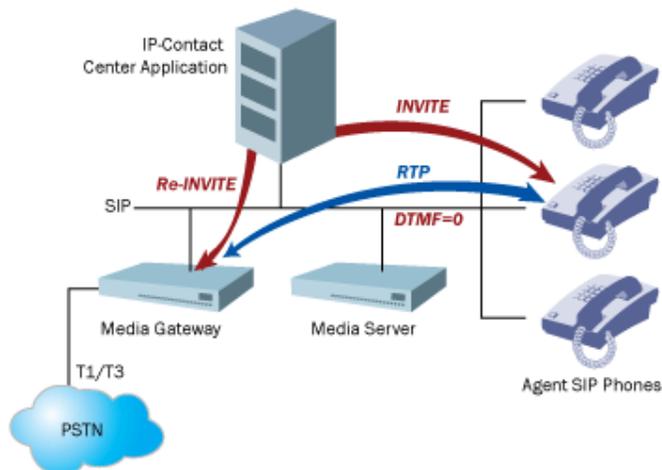


Figure 10: Inbound Call Flow - Step #5

The application then responds by choosing an agent to handle the call, sending an INVITE to the agent’s phone and a re-INVITE to the media gateway. This re-establishes the RTP stream and allows the conversation to proceed between the caller and the agent. At the same time, the application uses the caller ID information from the original INVITE to drive a screen pop on the agent’s desktop.

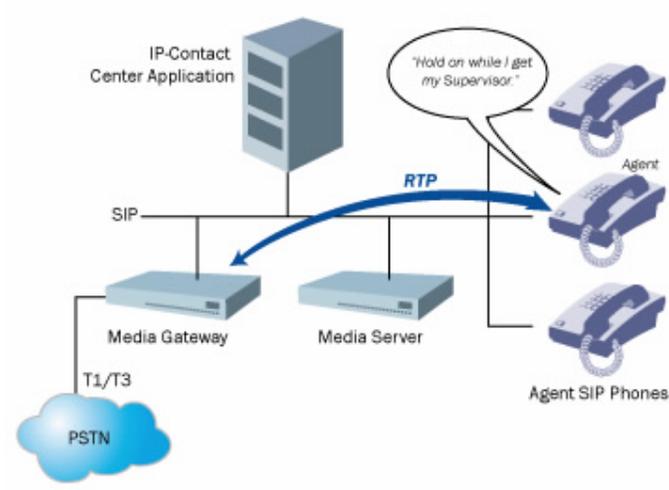


Figure 11: Inbound Call Flow - Step #6

During the discussion with the caller, a question comes up that is beyond the agent's skills, so the agent activates a consultation with a supervisor.

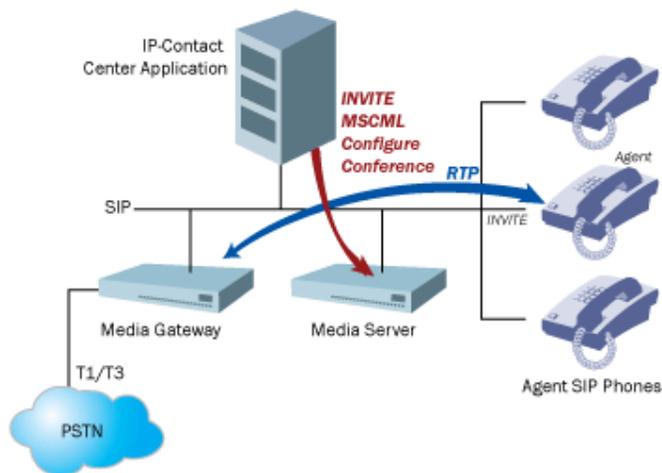


Figure 12: Inbound Call Flow - Step #7

To create a consultation, the application again selects a media server resource from the pool and initiates a MSCML Configure Conference.

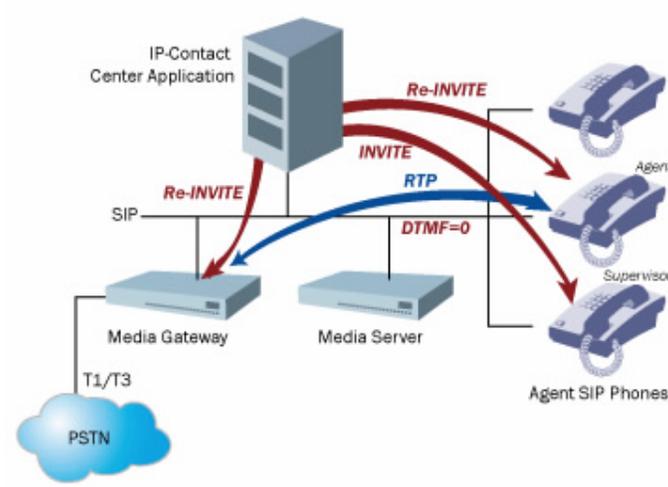


Figure 13: Inbound Call Flow - Step #8

The application then selects a Supervisor from the eligible pool and initiates a combination of INVITE and re-INVITE commands to the Agent, Media Gateway and Supervisor. This redirects the RTP streams to the newly created conference.

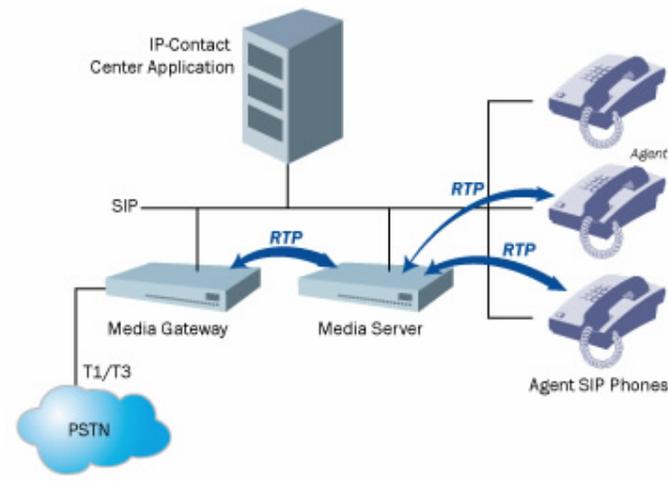


Figure 14: Inbound Call Flow - Step #9

With the Agent, Caller and Supervisor on the conference bridge, they then discuss the customer's question.

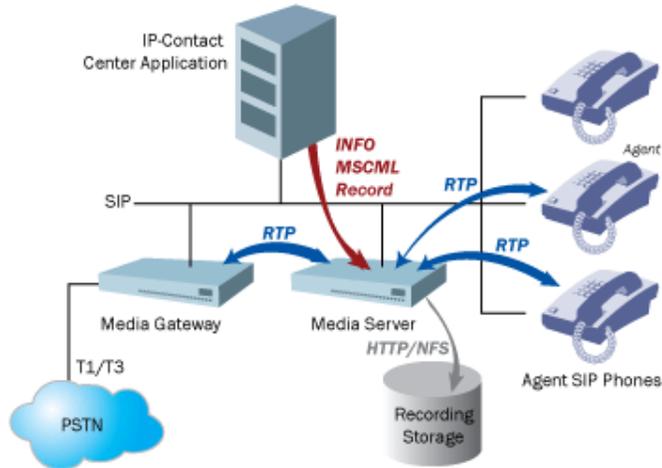


Figure 15: Inbound Call Flow - Step #10

To aid with future training of this agent and others, the Supervisor decides to record the call. The application initiates a MSCML Record operation on the conference, establishing a HTTP or NFS stream to bulk recording storage.

While highly simplified, the above example shows how complex call handling between media gateway, media server and agent phones can be accomplished with SIP and MSCML.

Sample Outbound Call Flow

More and more contact centers are starting to realize the value of “Blended Contact Centers”, mixing both inbound and outbound traffic. In other cases, contact centers need the ability to notify customers of flight schedule changes, special promotions or events, extended-credit conditions or a host of other proactive customer contacts. Making automated outbound calls requires the ability of the equipment doing the dialing to detect whether the call was busy, answered, or was answered by an answering machine or voicemail.

Continuing the example call-flow diagrams above, a series of diagrams follows showing an outbound call reaching an answering machine and leaving a pro-active customer message.

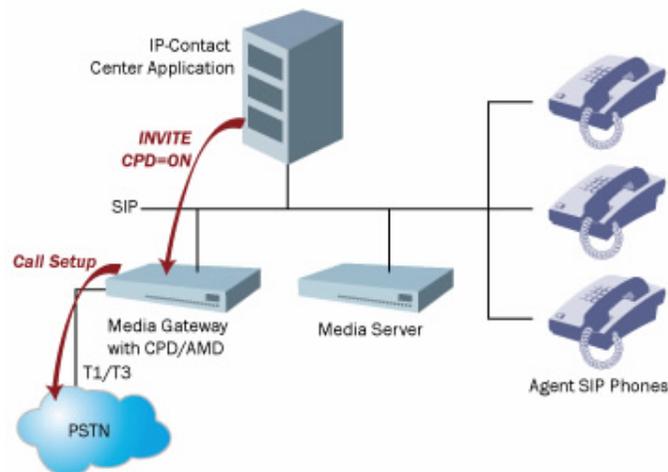


Figure 16: Outbound Call Flow - Step #1

The process starts with the application initiating an INVITE to the media gateway and activating the call progress detection features through optional tags found in the message. The media gateway then converts that request to an ISDN Call Setup message, dialing the number and waiting for the ISDN connect message. Once connected, the audio paths are connected and the media gateway starts listening for audible feedback on the status of the call.

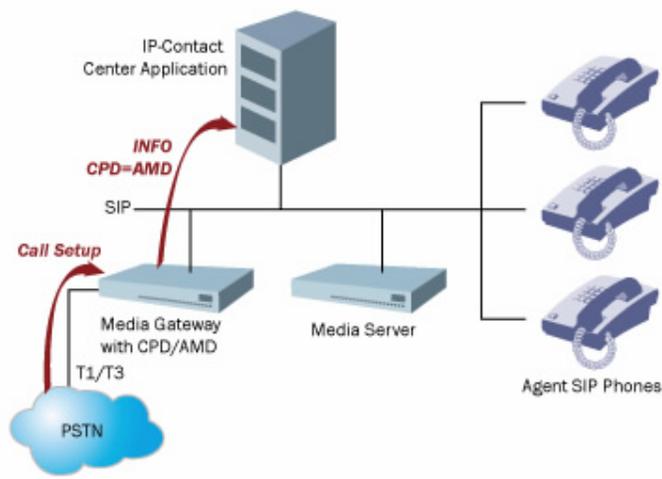


Figure 17: Outbound Call Flow - Step #2

Based on a set of sophisticated audio processing algorithms within the media gateway, an answering machine or voicemail is detected, and a SIP INFO message is sent to the application.

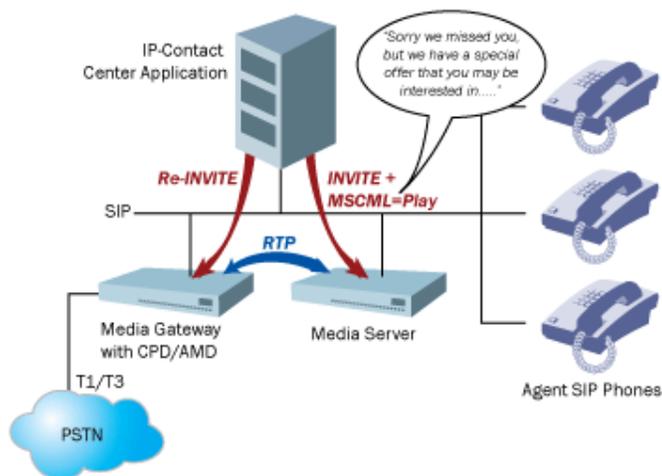


Figure 18: Outbound Call Flow - Step #3

Skipping a few steps that were covered earlier, the application would establish a RTP session between the media gateway and media server. Once connected, the application would then use the media server to play an announcement to the answering machine, in effect, leaving a message with promotional or notification details.

From this series of diagrams, you should be able to observe the power of having full call progress detection of an answering machine detection built-in to the media gateway. Integrating these features into the gateway reduces costs, speeds up call handling and simplifies the development of applications.

Migrating to SIP Trunking

A question you may be asking yourself now is “What about when the PSTN goes away?” The SIP architecture addresses this change and provides a smooth transition to an all-IP model.

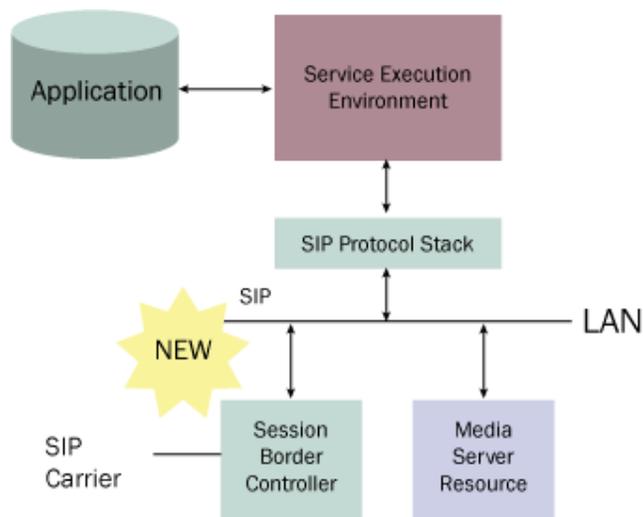


Figure 19: SIP Architecture for SIP Trunking

In deployments where a direct interface to a SIP carrier is available and there is no PSTN, the media gateway function is replaced with a Session Border Controller (SBC). The SBC provides important security and interoperability functions, including:

- Authentication
- Public/Private Network Address Normalization
- NAT Traversal
- Topology Hiding
- Session Admission Control
- Session-based Firewalling
- Encryption/Decryption
- Service Level Agreement (SLA) assurance
- Rouge RTP Detection and Deep Packet Inspection
- Protocol conversion (H.323 to SIP, or SIP to SIP)
- Protocol variant interoperability

Security

A common area of concern is confidentiality and security within contact center operations, especially within the Government, Financial Services and Health Care markets. Beyond the federally mandated regulations, a wide range of companies are only now beginning to realize their potential vulnerabilities and as a result, CIOs are requiring that their voice and data infrastructure be secure from end-to-end. This is of particular concern if the contact center is planning any remote WAHA, work centers or off-shore agents that would depend on non-secure data circuits to transport voice and data.

The SIP architecture addresses these needs with two key features:

SIP over TLS (SIPS) – using the well understood security features of Transport Layer Security (RFC 4346) are leveraged to secure the SIP messages, keeping the call control information private.

SRTP – in addition to securing the call control, encryption of the voice path is accomplished via Secure Real-time Transport Protocol (SRTP). SRTP (RFC 3711) provides confidentiality, message authentication, and replay protection to the RTP traffic and to the associated control traffic.

Embedding these protocols within the media gateways, media servers and session border controllers enables security from end-to-end and greatly improves contact center security.

AudioCodes Solutions

When designing or deploying SIP-based IP Contact Centers, AudioCodes offers a number of products that can help with your efforts.

Media Gateways

MediaPack Gateways– from as few as two FXO or FXS analog ports up to 24 FXS interfaces or BRI ISRN interfaces, AudioCodes MediaPack media gateway products provides complete turnkey analog gateway functionality. All MediaPack media gateway products include an onboard SIP protocol, making them ideal for us in the SIP Architecture.



Mediant Gateway Systems – from as few as one T1 span (24 ports) up to three DS3 interfaces, AudioCodes Mediant media gateway products provides complete turnkey digital gateway functionality. All Mediant media gateway products include an onboard SIP protocol, also making them ideal for us in the SIP Architecture.



TrunkPack – a family of blades that deliver media gateway functionality in a PCI or cPCI form-factor without device drivers or complex APIs. Starting at one T1 span up to three DS3 circuits, TrunkPack media gateways include on-board SIP protocols for use in the SIP Architecture.



Media Server Resources

AudioCodes IPmedia™ Server Platforms – deliver state of the art media processing and protocol technology in a single 1U form factor that is ready to rack and stack. Supporting both SIP and MSCML, the IPmedia line enables complex media processing needed for sophisticated applications.



IPmedia - a family of blades that deliver media resource functionality in a PCI or cPCI form-factor without device drivers or complex APIs. Starting at 30 sessions up to 2,016 sessions per board, the IPmedia resources include on-board SIP, NetAnn and MSCML protocols for use in the SIP Architecture.



Session Border Controllers:

nCite Session Border Controllers - provide secure traversal of firewall and network address translation (FW/NAT) systems, as well as denial of service (DOS) attack prevention through deep packet inspection at both the SIP signaling and VoIP media layers. nCite SBCs also provide end-to-end QoS, protocol Interworking, and support of high call processing volumes and over-subscription ratios required for a scalable residential VoIP service offering. AudioCodes nCite solutions offer proven interoperability with all major Softswitches, SIP servers, H.323 gatekeepers, call agents, application servers, media servers, media gateways, IP PBXs and numerous IP-based voice and video endpoints.



Summary

After reading this whitepaper, hopefully you can recognize how the power and flexibility of SIP and MSCML can be leveraged along with AudioCodes SIP products to build advanced contact center solutions.

SIP Contact Center Advantages:

- Greater Flexibility
- Put agents anywhere there is IP
- Local / Remote / WAHA Agents
- Greater Security
- Easier to Integrate
- ACD / IVR / Predictive Dialer all sharing hardware platform
- Easier to match customer needs
- Common COTS Application Servers
- Independent of expensive PBX equipment
- No Proprietary API Development
- More Cost Effective
- Faster time to market

About AudioCodes

AudioCodes Ltd. (NASDAQ: AUDC) provides innovative, reliable and cost-effective Voice over IP (VoIP) technology, Voice Network Products, and Value Added Applications to Service Providers, Enterprises, OEMs, Network Equipment Providers and System Integrators worldwide. AudioCodes provides a diverse range of flexible, comprehensive media gateway, and media processing enabling technologies based on VoIPerfect™ – AudioCodes' underlying, best-of-breed, core media architecture. The company is a market leader in VoIP equipment, focused on VoIP Media Gateway, Media Server, Session Border Controllers (SBC), Security Gateways and Value Added Application network products. AudioCodes has deployed tens of millions of media gateway and media server channels globally over the past ten years and is a key player in the emerging best-of-breed, IMS based, VoIP market. The Company is a VoIP technology leader focused on quality and interoperability, with a proven track record in product and network interoperability with industry leaders in the Service Provider and Enterprise space. AudioCodes Voice Network Products feature media gateway and media server platforms for packet-based applications in the converged, wireline, wireless, broadband access, cable, enhanced voice services, video, and Enterprise IP Telephony markets. AudioCodes' headquarters are located in Israel with R&D in the U.S. Other AudioCodes' offices are located in Europe, India, the Far East, and Latin America.

International Headquarters

1 Hayarden Street, Airport City
Lod, Israel 70151
Tel: +972-3-976-4000
Fax: +972-3-976-4040

US Headquarters

2099 Gateway Place, Suite 500
San Jose, CA 95110
Tel: +1-408-441-1175
Fax: +1-408-451-9520

info@audiocodes.com
www.audiocodes.com

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