



AudioCodes

WHITE PAPER

“Break Free” Leveraging SIP in developing enhanced applications

An examination of the constraints of legacy APIs in application development and an introduction to the SIP Architecture

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Introduction

Over the last 15+ years, literally thousands of enhanced applications have been developed for the legacy telecommunications infrastructure. From simple voicemail, to sophisticated contact center solutions, these Computer Telephony Integration (CTI) applications have built value on top of basic PSTN dial-tone, generating substantial revenue in both products and services. However, many of the CTI applications were developed using a restrictive and hard-to-learn architecture that limits the developer's choices in operating systems, choice of technology suppliers and incurs other serious restraints. This whitepaper will outline a migration strategy that leverages SIP to eliminate many of the past restraints and show how to "break free" from the bonds of the legacy CTI architecture.

The Demand for Applications

As the deployment of IP-based telecommunication systems migrate from a futuristic vision to reality, an important opportunity has opened up for legacy CTI enhanced application developers. This opportunity applies to the traditional messaging, call center, conferencing, recording, IVR and other specialized application solutions.

A significant lesson learned over the last few years by those "on the street" is that any serious telecom solution needs to include application offerings that closely match that customer's specific needs. There appears to be a void in the application space between the mature legacy CTI market and the new VoIP solutions. This void is a lack of specialized applications focused on specific market niches that the "one-size-fits-all" solutions currently on the market do not address.

No one will buy a next-generation IP-based telecommunications solution if it doesn't meet his or her business needs. For example: Would a hotel operator buy an IP-based phone system without a hospitality-enabled voice messaging system? No.

Just a few examples of applications that exist today in the CTI market that are hard to find for VoIP solutions include:

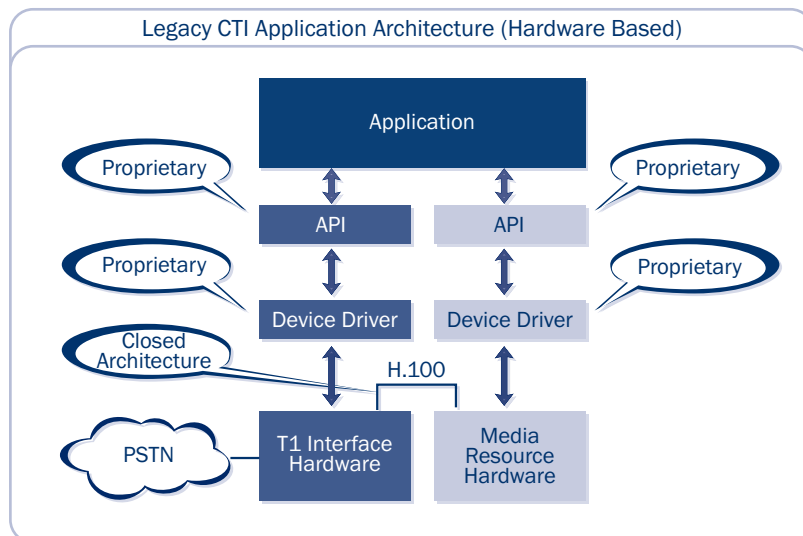
- Hospitality (hotels) messaging
- Professional office messaging and calendaring
- Call center quality recording
- Prison call management/recording systems
- Unified Communications and Fax
- and many more...

Most of these applications exist today, but only on legacy CTI platforms that aren't compatible with new SIP-based carrier or enterprise infrastructure. The result is a poorly integrated and cost prohibitive solution.

There is an opportunity for application developers to use this technology "disruption" as an opportunity to improve their position in the current market and create whole new markets with brand new applications.

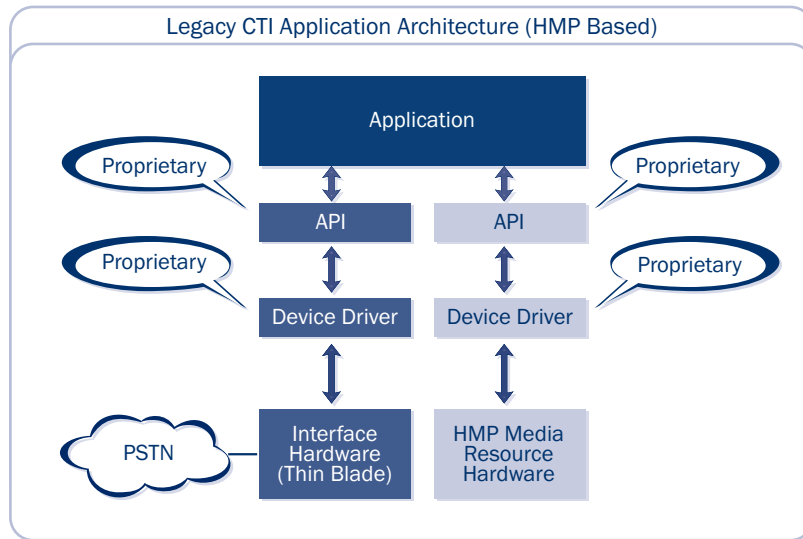
Legacy CTI Architecture

Since the inception of voice processing boards, developers have pretty much used the same architecture to build applications for the CTI market. A quick review of the architecture is below:



Most CTI applications support multiple DSP resource boards and/or interface boards as needed to connect to Loop Start, T1/E1, or Digital Station ports of enterprise or carrier TDM circuits. Solutions interconnect these different boards together with one of the popular TDM buses (H.100 or H.110).

With the advent of Host Media Processing (HMP), the architecture saw little actual change, the processing of the media just moved to an already over-taxed host CPU.



Both hardware and HMP solutions suffer from some very real shortcomings:

Proprietary APIs - Because the hardware and software components in this architecture originated from a number of different vendors, the APIs to control them are very different from one another. To work in this environment, developers generally have to spend a significant amount of time learning the different proprietary APIs and all their idiosyncrasies. Learning and maintaining these APIs is time consuming and creates "API job lock", limiting who can support or perform further development. "Job lock" is bad both for the engineer's career and for their employer.

Proprietary Device Drivers - Beyond the API issue, a key challenge of this architecture is that the device drivers provided by the vendors are tightly tied to the operating system. The net of this is that the board vendors will dictate which operating systems (and which versions of those operating systems) the developer can use. Shouldn't you be in charge of choosing your operating system?

Limited Packaging Options - because the CTI architecture depends on the API to communicate with boards or HMP software via device drivers, all the resources must reside in the same server. This requires specialized servers with numerous PCI slots, or special host processors that have been "certified" to support the real-time needs of HMP software. This seriously limits the choices of the server technology and as a result, limits features and drives up costs.

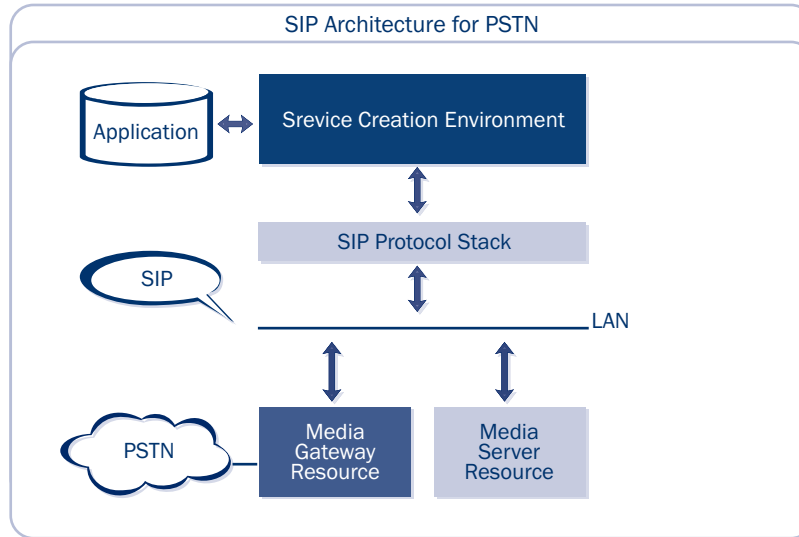
Monolithic Solutions - as a result of the above packaging challenges, applications built on the legacy CTI architecture are monolithic in nature. Any change to any one of the hardware or software parts requires the entire solution be taken off-line for service or upgrades. This forces the applications to be installed in each remote site and makes field maintenance very difficult.

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

Pace of Change - because of the complexity, modifications to applications based on APIs is very difficult and requires significant effort to completely test and debug development issues. This results in resistance to change and slows the evolution and innovation involved in new applications leaving the door open for more agile competition that can adapt quickly.

SIP Architecture

The SIP Architecture approach solves 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:



Application/Service Creation Environment – provides an environment for development and execution of the logic to be implemented by the completed application. Development tools include C/C++, Java, scripting languages, VoiceXML, CCXML and “drag and drop” graphical application environments.

SIP Protocol Stack - Common to this architecture is the use of a SIP protocol stack to communicate with the other resources in the architecture. Depending on the environment and the complexity of the application, the SIP protocol stack may be licensed, open source or developed in-house.

Media Gateway Resource – Provides connectivity between the application and the outside world. This can be via T1/E1 circuits, SS7, analog or any other media format required by the particular installation or application. Media gateways generally require dedicated hardware to interface to the appropriate physical interfaces outside the application. SIP is the dominant control protocol for Media Gateways, providing a standard and open control mechanism.

Media Server Resource – Responsible for all media processing including prompt playback, recording, tone detection, conferencing, transcoding or other complex media activities. The media server function can be separate hardware, boards or pure software – the actual form factor is very flexible. The key is that SIP and companion media processing protocols NetAnn or MSCML are used to manage the media server operations.

Advantages of the SIP Architecture

The SIP Architecture has a number of advantages over the legacy CTI architecture:

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. This reduces integration and development time, speeding time to market. As an added bonus, developers gain industry-wide expertise which makes staffing much easier.

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. SIP uses the TCP/IP stack built in to virtually all operating systems. Now the application developer gets to choose the operating system, not the software or board vendors.

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 - now the choice of packaging accommodates the end-customer's needs. This flexibility offers opportunities to optimize equipment sizing, accommodate maintenance requirements, while minimizing costs.

Distributed Solutions – because the SIP architecture utilizes IP for command and control between the various functional blocks, those individual blocks don't have to reside in the same physical site. Media Gateways can move to the edge. Application and Media Servers can be located centrally if needed. The key is that the solution designer now has many more choices.

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 scale down over a very broad range. The functional blocks are essentially “racked and stacked”.

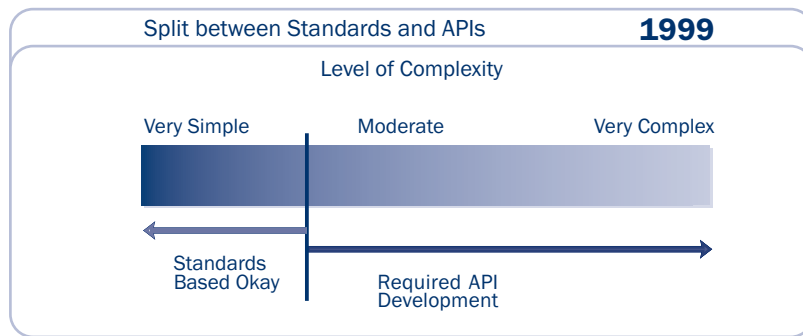
Agility – because the SIP architecture is more modular, application modifications can be implemented and tested more quickly, eliminating many of the exhaustive cycles of testing and debugging. The reduced effort allows developers to implement and deploy new applications more quickly and keep up with the customer demand.

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

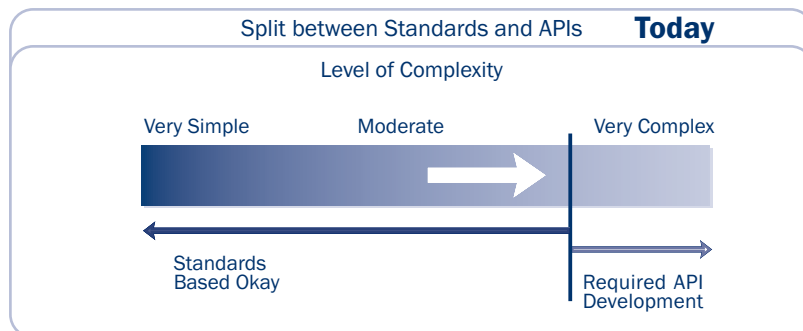
Shift in Development Complexity

A common question that arises is: “My application uses complex API features, can SIP address these needs?” Naturally, a detailed analysis would provide a concrete answer, but a trend is definitely developing in the marketplace that allows many more applications to leverage standards-based protocols and SIP.

All applications fall along a spectrum of complexity – some are very simple, others are very complex, most are somewhere in-between. Even as recent as 1999, the standards-based interfaces were very limited in capabilities and only the simplest applications could use standards and avoid using APIs.

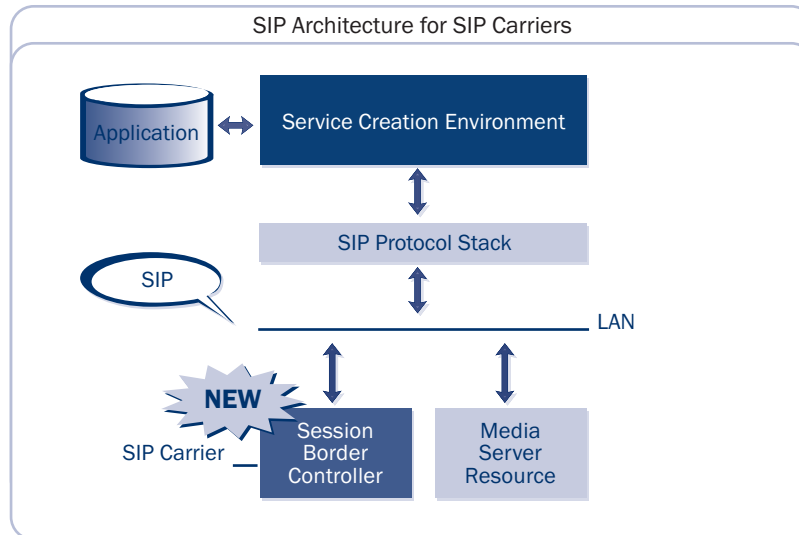


Skip ahead to today and the increased capabilities and features of SIP and the associated media control protocols has moved the cross-over line significantly to the right. As a result, many more applications will be able to leverage the SIP architecture and capitalize on the associated benefits.



Migrating to an All-IP Solution

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.



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

AudioCodes Solutions

If you have an application and are developing your plan to migrate it to support VoIP, AudioCodes offers a number of products that can help with your migration 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 use 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 use 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

Hopefully this paper has helped you to understand that there is a tremendous opportunity to fill the application void in the VoIP technology space and the advantages of leveraging the SIP architecture in migration of your application. Using this information as a guide, developers can now choose the approach and products that makes the most sense for your application and break free from the restraints of the legacy CTI architecture!

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

Contact us: www.audiocodes.com/info

Website: www.audiocodes.com

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