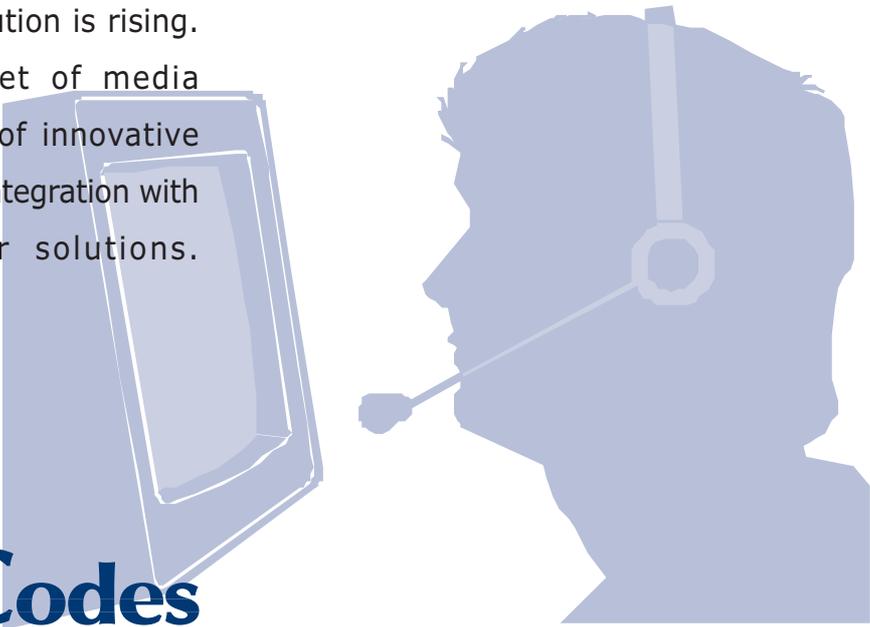




Application Note

Deploying a Call Center Solution

As the market place for Call Centers continues its rapid growth, a demand for leveraging the inherent benefits of VoIP for deploying a smart, distributed and cost-effective solution is rising. AudioCodes' comprehensive set of media gateways, equipped with a set of innovative features, is the perfect match for integration with leading industry Call Center solutions.



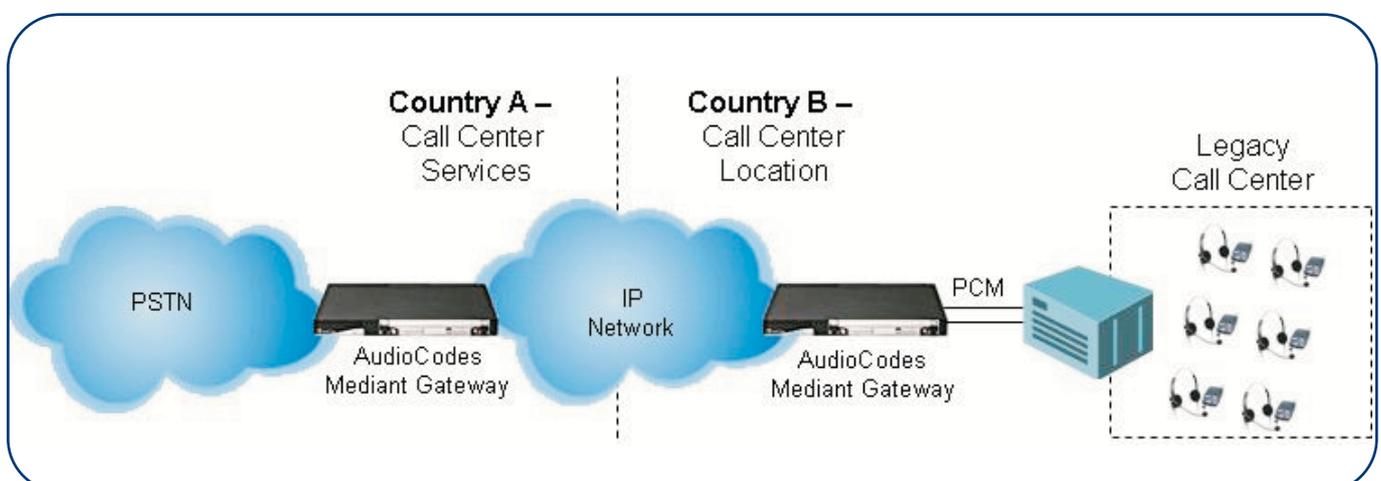
An increasing number of enterprises are utilizing VoIP to deploy distributed Call Centers in order to benefit from reduced costs and cheaper yet skilled labor outside of their marketing territory. Others are using VoIP in order to augment existing Call Centers or to create “virtual” Call Centers in which call representatives are situated at various geographical locations, but are still providing a unified customer experience for callers. IP-based call centers are gaining popularity as enterprises that have already installed IP-PBXs for their general telephony needs start to add call-center functionality. In addition, service providers are deploying hosted Call Centers in which small businesses can provide their customers with the appropriate attention, without owning equipment.

The Mediant and MediaPack gateway families boast a wide set of features which allow them to be deployed, in conjunction with legacy and leading IP-based Call Center Application Servers, into various network solutions. The following list highlights some of the applicable features of the AudioCodes gateways:

- Superior voice quality with reduced bandwidth codecs for excellent service quality even in challenging network conditions, such as over the open Internet
- On-board dialing plan including DNIS-based and source-based routing, traffic load-balancing, and alternate routes, eliminating the need for a routing server
- Support for DNS server lookup for integration with existing IT systems
- Caller ID (ANI) presentation for preserving ACD functionality
- T.38 support for enabling fax communications with the Call Center
- Answering Machine Detection (AMD) and Special Intercept Tones (SIT) detection for optimal operation and effective resource management with predictive dialers
- ThroughPacket™ technology for RTP header optimization allowing reduced bandwidth requirements
- Support of RFC 2833 and SIP Info method for carrying out-of-band DTMF signaling, thus guaranteeing continued IVR operation
- High performance for peak hour service

Out-of-Territory Call Center

Connecting over IP to a legacy Call Center in another country is easily achieved by deploying AudioCodes Mediant gateways, illustrated in the figure below:

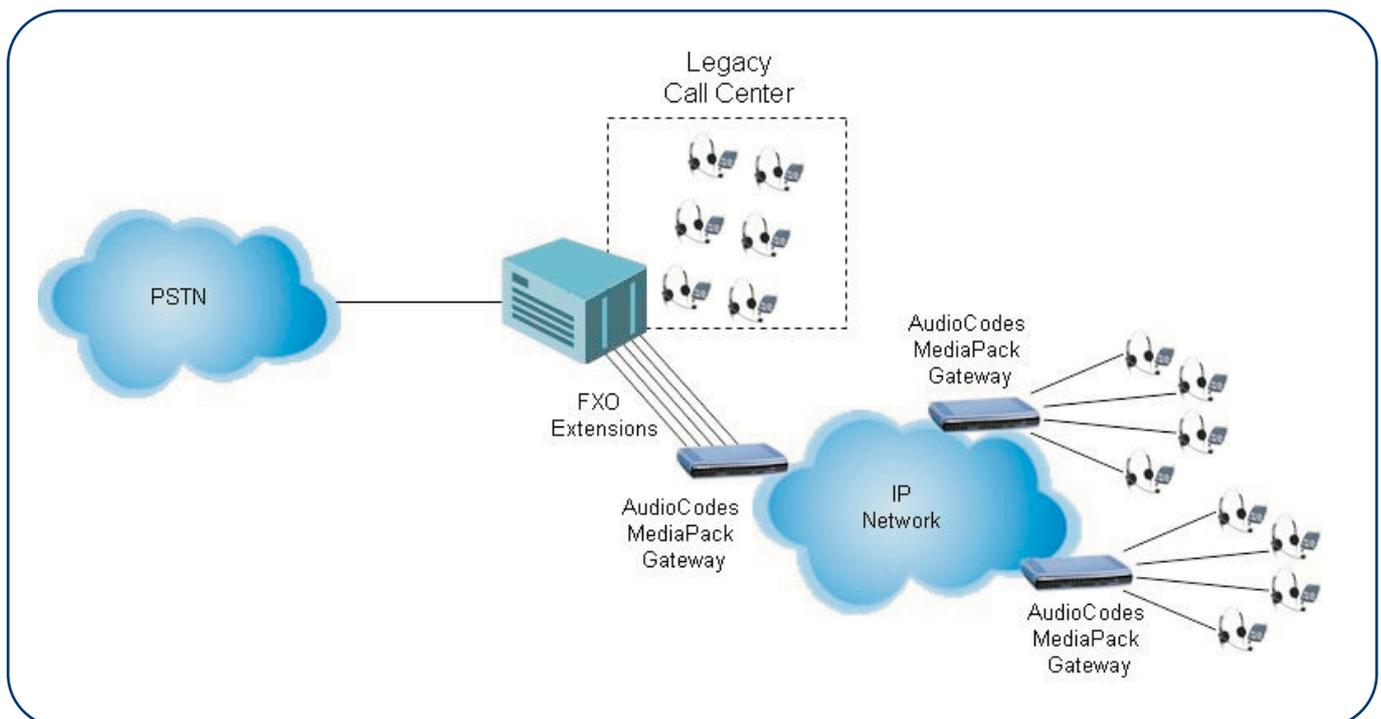


This type of deployment allows AudioCodes Mediant gateways to connect to the PSTN in country A using PRI interfaces, and to the legacy Call Center in country B using PRI/CAS signaling.

Utilizing the flexible on-board dialing plan in the origination gateway, calls are routed to a termination gateway. Note that in this process, it is possible to include additional origination or termination gateways that can connect to various Call Centers. Furthermore, it is possible to connect numerous Call Centers to a single termination gateway. In this type of deployment, a service call to a Call Center in country A is routed over IP to country B and directly to the selected Call Center.

Distributing an Existing Call Center

Increasing the number of distributed attendants to an existing Call Center becomes an approximate plug-and-play task using AudioCodes versatile set of gateways which support both FXS and FXO analog extension interfaces.

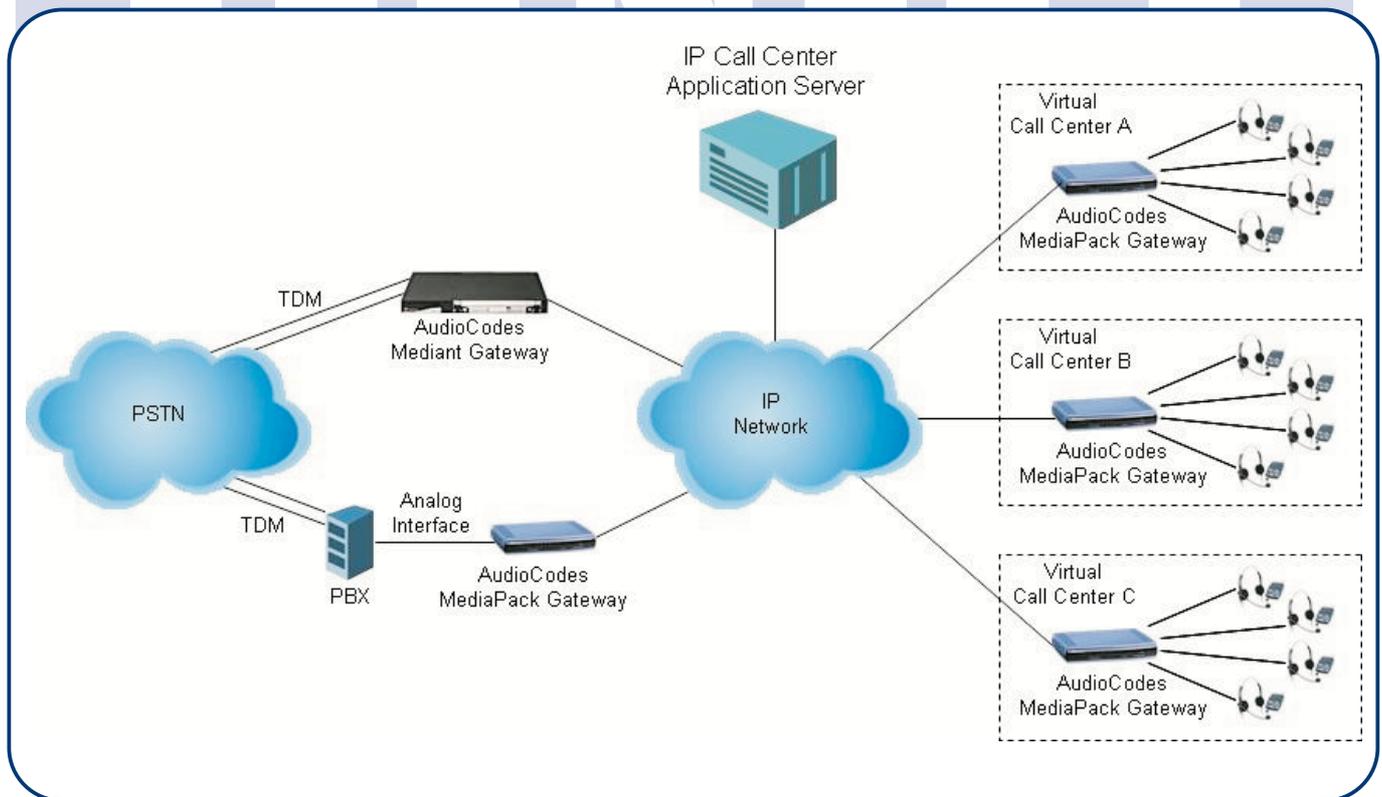


By connecting a MediaPack gateway with FXO extensions to the Legacy Call Center it is possible to locate the extended attendants in remote locations, while maintaining the same Call Center capabilities including IVR and ACD functionality. The MediaPack gateways on the remote side connect to analog extensions using FXS interfaces, thus allowing attendants to work from alternate locations which allow “follow-the-sun” 24-hour service, cost reduction, survivability and service continuity in cases of disaster recovery. AudioCodes’ gateways support call hold and call transfer allowing call handover and consultation, and an internal announcement server for playing “music-on-hold”.

Hosted All-IP Call Centers

As operators and service providers seek to increase revenue from hosted services, the hosted All-IP Call Center segment - a lucrative application - is being pursued. AudioCodes has completed interoperability with leading All-IP Call Center Application Servers, facilitating a deployable solution within a short timeframe.

Using the MediaPack gateways with FXS extensions, it is possible to connect remote Call Center attendants, who are virtually located within the IP Call Center's application



server and who are managed by it. These extensions are included in the ACD engines of the application server and the caller's experience is unaffected. A tiered architecture of Virtual Call Centers can be created in this manner, allowing operators and service providers to host numerous Call Centers over a shared infrastructure. In addition, optimal performance for maximal concurrent call usage is achieved using Mediant gateways' support for various call transfer mechanisms, including RLT, TBCT, and ECT which enable expensive trunk side and DSP resources to be released when calls are transferred out of the Call Center.

AudioCodes gateways have been tested and certified by leading vendors of IP Call Center solutions. For the comprehensive and updated list please visit our web site http://www.audiocodes.com/Content.aspx?voip=2019#Contact_Center. For more information please contact your local representative or an authorized AudioCodes dealer.

Ref. # LTRM-09003 03/06 V.1