Telurix HD Audio Conferencing Solution

The Telurix Communications Server (TCS) is the industry's most flexible, scalable, and cost-effective SIP-based VoIP HD Audio conferencing platform. TCS is a software-only solution that runs on low cost, off-the-shelf Intel-based servers. It offers communication service providers and application developers an impressive amount of functionality at an unprecedented price / performance ratio.

TCS is a feature-complete SIP signaling, media processing, and web application server. TCS implements call routing, IVR, bridging, conferencing, and transfer functionality − all in a single application suite. TCS integrates with Web-based and SOAP infrastructures, acting as both an HTTP server and an HTTP client.

With TCS, your development team can rapidly build a customized conferencing solution to your exact market requirements while focusing on the intent of the application. TCS will manage all the underlying complexity.

Programmable

TCS is controlled by scripts written in a proprietary XML-based language (CSXML) or in industry standard VoiceXML 2.1. These scripts can be generated by applications running on Web servers, allowing your web development team to use the application servers of their choice to develop conferencing applications to your requirements. The CSXML scripting language is ideally suited for the AJAX programming model, enabling higher application performance by using static scripts with dynamic data.

Extendable

TCS is modular and can be extended to provide advanced IVR and call control functionality. Custom media processing, such as GSM noise removal or echo cancelation, can be provided via extension modules. Extension modules can also be used to enable additional communication methods, such as RADIUS or to integrate with third party TTS, ASR, and voice authentication engines, such as LumenVox, Nuance, or NeoSpeech.

Scalable

By scaling across multiple standard servers, TCS is easily expanded to handle thousands of concurrent calls with the addition of the hardware of your choice. A single modern Intel Dual-CPU Xeon based server will support up to 2,000 concurrent VoIP calls.
Each TCS server can provide both signaling and media processing functions, allowing for very flexible deployment scenarios. TCS can be deployed so that all servers are identical and perform the same functions or as a hierarchy, where some TCS servers are responsible for call control and other TCS servers are dedicated to media processing.

Scalable software for HD Audio Conferencing for Intel-based platforms

Feature Complete

TCS provides all of the SIP signaling, media processing, and IVR functionality necessary for a conferencing application. In-bound and out-bound calling functionality, IVR menus, mixing, SIP registration processing, and call recording can be implemented using TCS. Furthermore, TCS supports server side NAT Traversal and can act as a STUN server. No additional session border controllers or load balancers are needed.

Reliable

TCS provides automatic and intelligent load balancing of conferences across all available media resources in your array. Should a media resource fail, TCS will continue to process conferences on the remaining resources, insuring uninterrupted services. TCS supports the addition of new resources while in service, so that you can expand with no downtime.

Interoperability

TCS has been tested and deployed with the Level 3, Verizon, Telia, Qwes and works with a large number of SIP endpoints, including phones, gateways and PBXs from Cisco, Sonus, Polycom, Linksys, Audiocodes, and Asterisk.

Reference Applications

TCS ships with a standard meet-me audio conferencing application as well as a series of sample ‘getting started’ applications. The system is designed for customization to your requirements. Whether you’re looking for a simple reservation-less conferencing system, sophisticated event management system or specialized conferencing application, TCS can provide the solution.

Sample Current Customers

Customer ‘A’ offers HD Audio Conferencing for business with a unique twist of features as its primary competitive advantage, allowing for pin-less conference entry and out calling to participants. This is unlike services offered by traditional audio conferencing providers. This service can host up to 400 participants in a single conference.

Customer ‘B’ is a financial services company using TCS for automated debt collection. Using IVR, conferencing, and the TCS web API, its programmers quickly developed an application to call consumers, play various informational messages, collect input from the consumer and conference a ‘live’ agent onto the call. The customer is currently making 500,000 outbound calls a night.

Customer ‘C’ developed an emergency notification service using TCS. In the event of an emergency, TCS will automatically dial various parties based upon our web API and place them into a conference. The platform was further customized to make multiple re-tries and ignore voicemail systems.

Professional Services

Most customers can deploy the TCS system themselves. But if you need additional customization or just a helping hand to get started, Telurix Professional Services can work with you to deliver a custom solution.

Pricing

The TCS platform, including the base system and sample applications, is available today. The base system supports 250 concurrent calls. Additional licensing for 250, 500 and 1,000 concurrent calls is also available. Contact us for a custom quote.

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www.telurix.com

Software Specifications

System Software Overview
• SIP Application Server
• Media Processing Server (Software DSP)
• Web Server and HTTP Client
• Telurix Scripting Language (CSXML)

SIP Signaling
• Complete control over SIP signaling
• Supports calling features
• Built-in back-to-back SIP user agent (B2BUA),
• Supports non-call-control SIP applications

IVR
• Play, Record, DTMF detection and generation
• VXML 2.1 Support (plus CSXML)
• Optional third-party TTS and ASR engine integration

Audio Mixing
• Multi-party, n-loudest with VAD
• Listen-only, play-only, whisper

Session Border Controller
• Server Side NAT Traversal / STUN
• Integrated High Performance Media Proxy
• Signaling and Media isolation

Web and SOAP Integration
• Built-in HTTP client
• AJAX based programming model
• SOAP and REST data services
• HTTP call control

Sample Conference Application
• Meet-me Conference
• Various Conference mode
• Conference start modes
• Entry/Exit chimes
• Name record/announcement
• Music on hold
• Caller count
• Record call
• Outbound Calling
• Multi-Tenant Support

Participant Capabilities
• Change mode of conference
• Mute / unmute own line
• Lock / unlock conference
• Roll call / count of participants
• Joining announcements on/off
• Start / stop recording
• Force end of conference

Reporting
• Current activity report
• CSV Call Detail Record Generation
• RADIUS interface
• Push call results via HTTP/REST/SOAP

Recording
• On demand recording (start/stop/pause)
• Conference recording in .mp3 / .wav format

Security
• Long conference duration auto disconnect
• Auto disconnect for invalid access
• Host can terminate call

Live Conference Manager
• Current Conference Status
• Full conference control
• Full participant control

Technical
• Load balanced across multiple servers
• Initial support for 250 party conference
• Base platform 2x1U Server, 2,000 concurrent ports
• Performance of 100 CPS
• Auto-fail over of all conference processes
• Multiple CODECs are automatically transcoded

Support
• Extended business day support
• On-call response to support (out of business hours)

Codecs
G.711, G.722, AMR-WB (beta), SILK (beta)
SIP, RTP, RFC 2833 DTMF

Hardware (recommended)
Intel E-series Xeon Dual CPU or better server, 8 gigabytes memory, RAID 1 disk subsystem

Operating Systems
Production: Linux (Redhat, CentOS, Debian, Ubuntu)
Development: Windows

Availability
Now, downloadable trial available

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