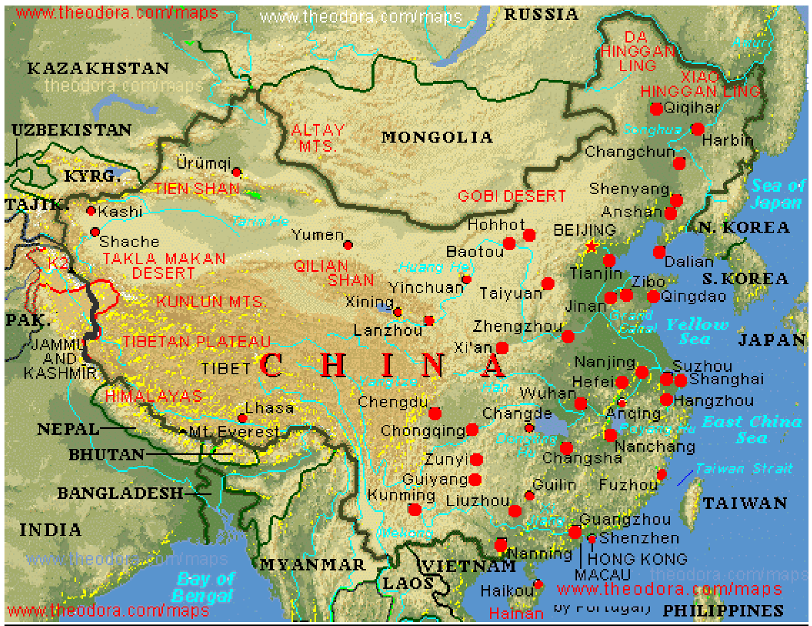
Across China: The State Information Center's Nationwide Voice-over-IP Network



A customer case study from Quintum Technologies

The State Information Center (SIC) is the government body responsible for delivering information technology services throughout the People's Republic of China. The SIC has regional offices and information centers dispersed throughout the country's 30 provinces and 16 large cities, operating an advanced nationwide private IP data network running over an ATM backbone. As part of an extensive upgrade of its communications facilities, SIC decided to link its 500 regional offices and Beijing headquarters using voice over IP (VoIP).

The SIC's long-term objective is to develop an innovative model for the government to implement VoIP. They intend to connect all government information centers using VoIP as part of its bold "e-Government" initiative. By using VoIP, SIC will be able to cut costs on long distance calls by having their employees utilize their intranet instead of China's long distance telecom company. This will also increase the security of their phone conversation by using the voice VPN.

The Challenge: Building the SIC VoIP Network

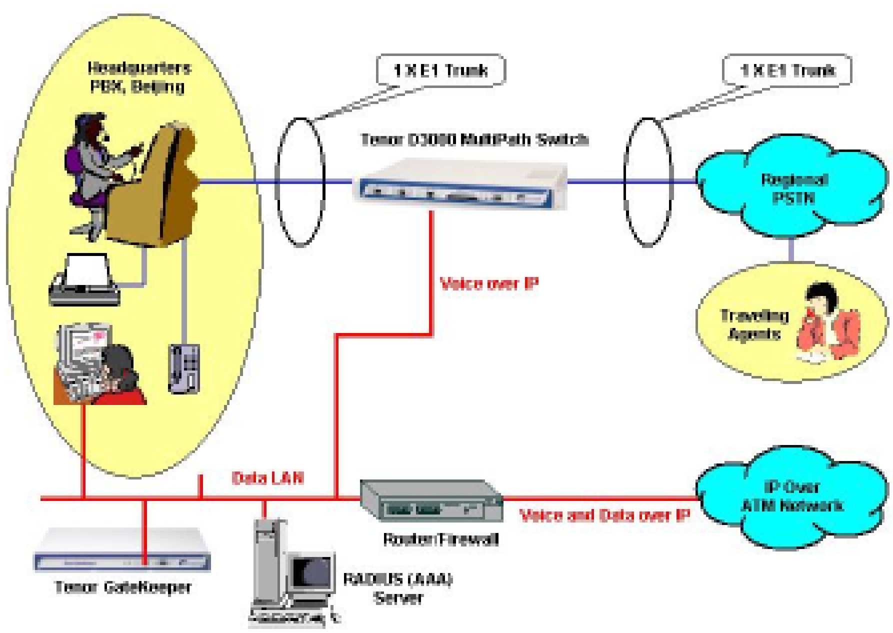
Early on, SIC technical managers developed a core set of requirements for its VoIP architecture. These requirements included:

- Excellent voice quality
- High reliability – including an effective form of backup for the system in the event of any problem with the IP network
- The ability to “hop on” and “hop off” the VoIP network from the public switched telephone network (PSTN)
- Optimum voice compression to ensure that VoIP service could be delivered even to locations with limited bandwidth

After an extensive evaluation of available VoIP products, SIC selected Quintum Technologies as its VoIP vendor. This decision was based on the advanced features and high voice quality provided by Quintum’s Tenor VoIP MultiPath Switches, as well as their support for both analog and digital PSTN connections.

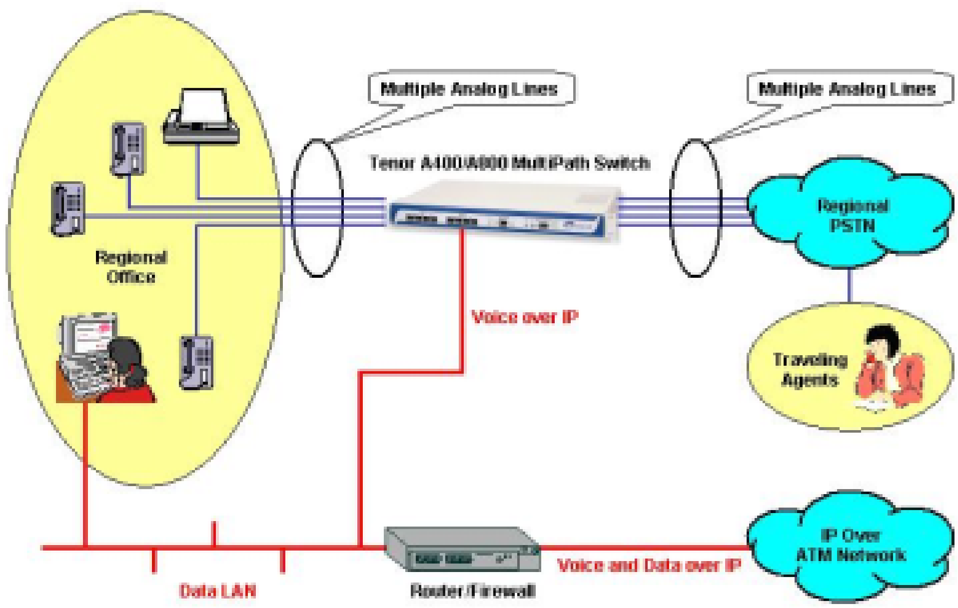
Of particular importance was Tenor’s patented SelectNet™ intelligent switching capability, which ensures that voice communications won’t be adversely affected by problems in the IP network. In the event of such problems, the Tenor switches can automatically re-route calls over the PSTN – allowing even active calls to continue without interruption.

The Tenor switches connect to both the IP network and the PSTN. The analog switches in each regional office support either analog phones and fax machines or analog PBX systems and connect to the local PSTN via analog lines. The digital switch at SIC’s Beijing headquarters connects to the local PSTN via E1 Primary Rate ISDN lines (see following diagrams).



Beijing Headquarters Configuration

THE SOLUTION: SIC initially purchased 49 Tenor A400 four-port Analog VoIP MultiPath Switches, and one central Tenor D3000 thirty-port Digital VoIP MultiPath Switch. A second Tenor D3000 was later installed to provide redundancy at the central site. Additional A400 and A800 eight-port analog switches have also been purchased for expansion of the system. A standalone Tenor GateKeeper unit will be added to support anticipated traffic growth.



Regional Location Configuration

How It Works

SIC's implementation of Quintum's Tenor VoIP solutions resulted in a highly effective and flexible communications solution. The architecture met all of the agency's requirements both for functionality and performance.

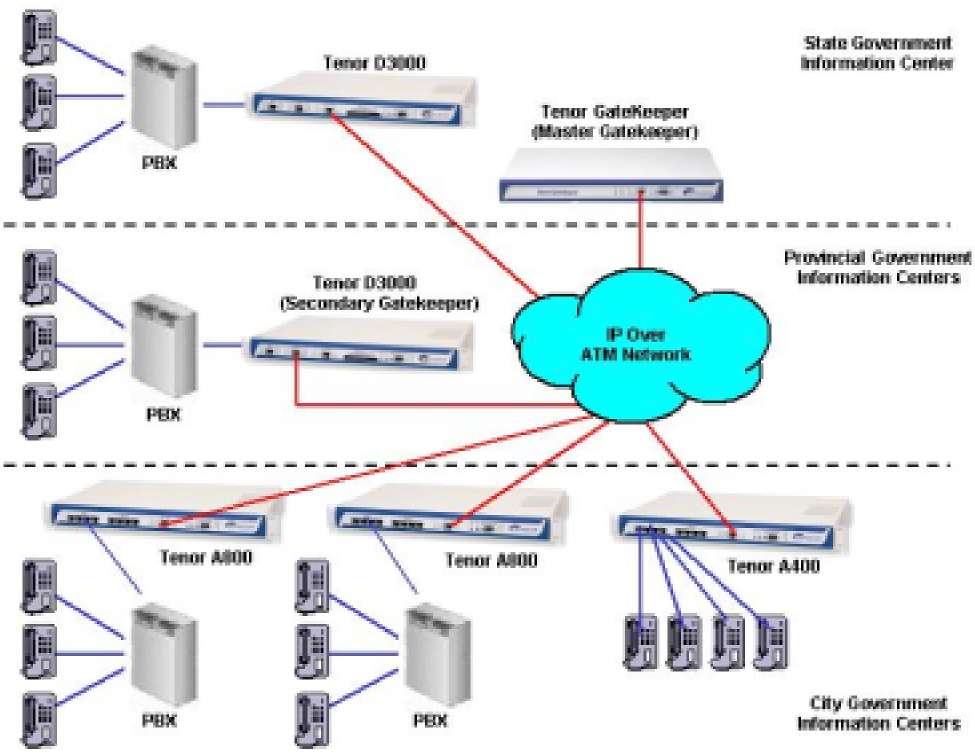
Inter-office calling

With the Tenor-based VoIP system in place, agents in any VoIP-enabled SIC location can place calls to any other VoIP-enabled location over the IP network. The Tenor's MultiPath design couples it tightly with both the telephony network (i.e. the PSTN) and the data network. The Tenor's built-in call routing intelligence enables it to route calls from agents to either the PSTN or the IP network, based on numbers dialed.

Scalable, reliable "gatekeeping"

The SIC VoIP network uses Quintum's intelligent multi-tier gatekeeper architecture to distribute gatekeeping duties between a master gatekeeper and secondary gatekeepers at the SIC headquarters and a master gatekeeper at the provincial information centers. This distributed, hierarchical architecture shares the load between the gatekeepers, resulting in maximum efficiency and reliability.

The Tenor GateKeeper, which acts as the master gatekeeper, controls the entire network, which is divided into individual zones for each province. Each zone is controlled by the secondary gatekeeper built into the provincial Tenor switches. The Tenors deployed in city government information centers within the province all register with this secondary gatekeeper. The secondary gatekeepers, in turn, register with the master gatekeeper, which acts as the central controller for the complete network. The master Tenor builds a central database and passes information about the complete network to each individual gatekeeper, so that each gatekeeper can route calls to any endpoint on the network.

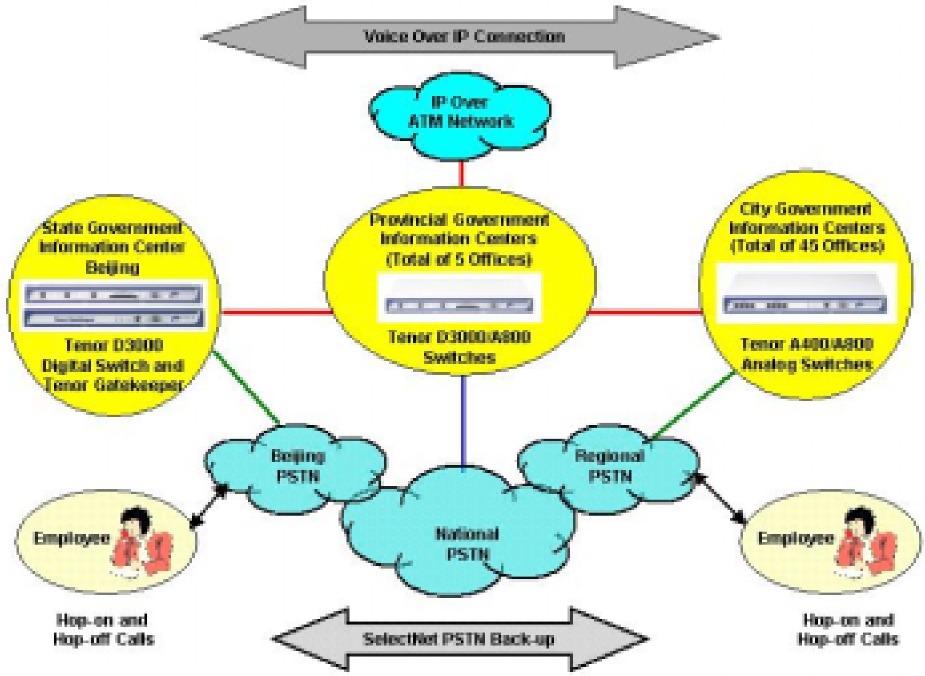


Hierarchical Gatekeeper Architecture

"Hop on"/"Hop off" Capabilities

One of the important considerations in the choice of equipment was support of SIC agents traveling throughout China. It was vital for these agents to be able to make calls to any of the Regional Offices or to the headquarters in Beijing over the VoIP network.

This capability to "hop on" to the network is also provided by the Tenor's MultiPath architecture. Just as the Tenor is able to route calls from local users to either the PSTN or the IP network, calls coming in to the Tenor from the local PSTN can also be routed either to the local office or the IP network, based on the number dialed. Traveling agents can thus call in to the nearest Regional Office via the PSTN and "hop on" to the SIC VoIP network to call any other office on the network.



"Hop on"/"Hop off" Capabilities

Dial-in access to the VoIP network is provided by a two-stage dialing capability built into Tenor switch. An agent wishing to access the VoIP network dials in through the PSTN to a number designated for "hopping on." The Tenor answers the call and responds with a secondary tone. The agent then enters a PIN code. After verifying the validity of the PIN code, the Tenor presents the caller with a second dial tone – enabling them to dial the destination they wish to reach.

This two-stage dialing process is made even more secure by using the Interactive Voice Response (IVR) and RADIUS server interface feature of the Tenor. In this case the Tenor interfaces over the IP network with a central Remote Access Dial-In User Service (RADIUS) Server, which carries out the caller Authentication, Authorization, and Accounting (AAA) processes. The Tenor leads the caller through the whole procedure using voice prompts and passes the appropriate information to the RADIUS Server for highly secure access control.

Calls coming in to a Tenor via the SIC VoIP network can also be switched out to the local PSTN based on the dialed number. Traveling agents can thus also place calls across the country over the IP network by "hopping on" to a nearby SIC office, having the "long distance" segment of the call travel over the SIC IP network, and then "hop off" the network at the office nearest their desired destination and onto the local PSTN to complete the call. Agents can thus make full use of the SIC network to inexpensively call anywhere within China.

"Auto-switched" PSTN Back-up

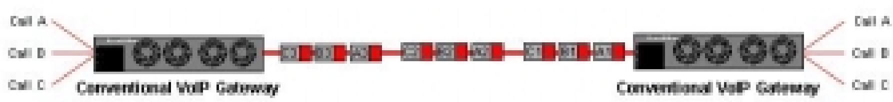
A major concern of SIC in implementing VoIP was maintaining reliable, high quality voice communications in the event that its IP network experienced problems such as congestion or delay. This concern was eliminated by Tenor's SelectNet technology.

SelectNet technology constantly checks the quality of the IP connection between Tenor switches during VoIP calls and, if necessary, automatically launches a back-up connection over the PSTN. When a call is placed over the IP network between two Tenors, the two Tenors constantly monitor the status of their IP connection, based on parameters such as packet loss, delay, latency, and jitter. If the SelectNet algorithm senses that the connection has become inadequate for maintaining high voice quality, the originating Tenor launches a back-up call over the PSTN. Once this PSTN connection is established, the call is switched in real-time from the IP network to the PSTN. The switch-over is transparent. SIC thus gains complete protection against IP network issues affecting its vital voice services.

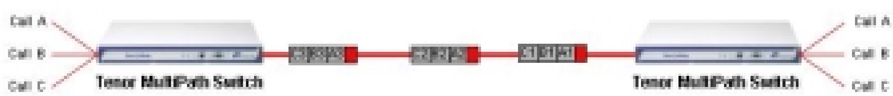
Bandwidth Conservation

Because access bandwidth is as low as 64 Kbps at some locations, SIC needed its VoIP system to be highly bandwidth-efficient. Fortunately, Tenor switches feature Quintum's exclusive PacketSaver™ packet multiplexing technology, which greatly reduces the amount of bandwidth required to support multiple calls flowing between the any two SIC offices. PacketSaver routinely achieves bandwidth savings of as much as 50%, resulting in per-call requirements of a little as 6Kbps per call.

PacketSaver achieves these savings by aggregating samples from multiple VoIP conversations and packing them into a larger IP packet with a single IP header. This dramatically reduces the "overhead" generated by adding a bulky IP header onto every individual voice sample.



Conventional VoIP Transmission Sends Many Redundant Packet Headers



Tenor MultiPath Switches Use PacketSaver™ to Minimize Bandwidth Usage

Quintum's Tenor: Powerful, Practical VoIP for Large-Scale Networks

11 months into its VoIP implementation, SIC has found that Quintum's Tenor products deliver an economical, sophisticated, and highly reliable solution for enabling quality voice communications over its existing IP network infrastructure. Tenor's intelligent switching provides flexible on- and off-network call capabilities, highly efficient bandwidth utilization, and scalable gatekeeping functionality. Just as importantly, its auto-switching ensures that users will always be able to complete their calls – regardless of momentary problems on the IP network. Tenor switches have thus proven themselves as the ideal platform for building the cross-China VoIP network that SIC required as part of its larger e-Government strategy.

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