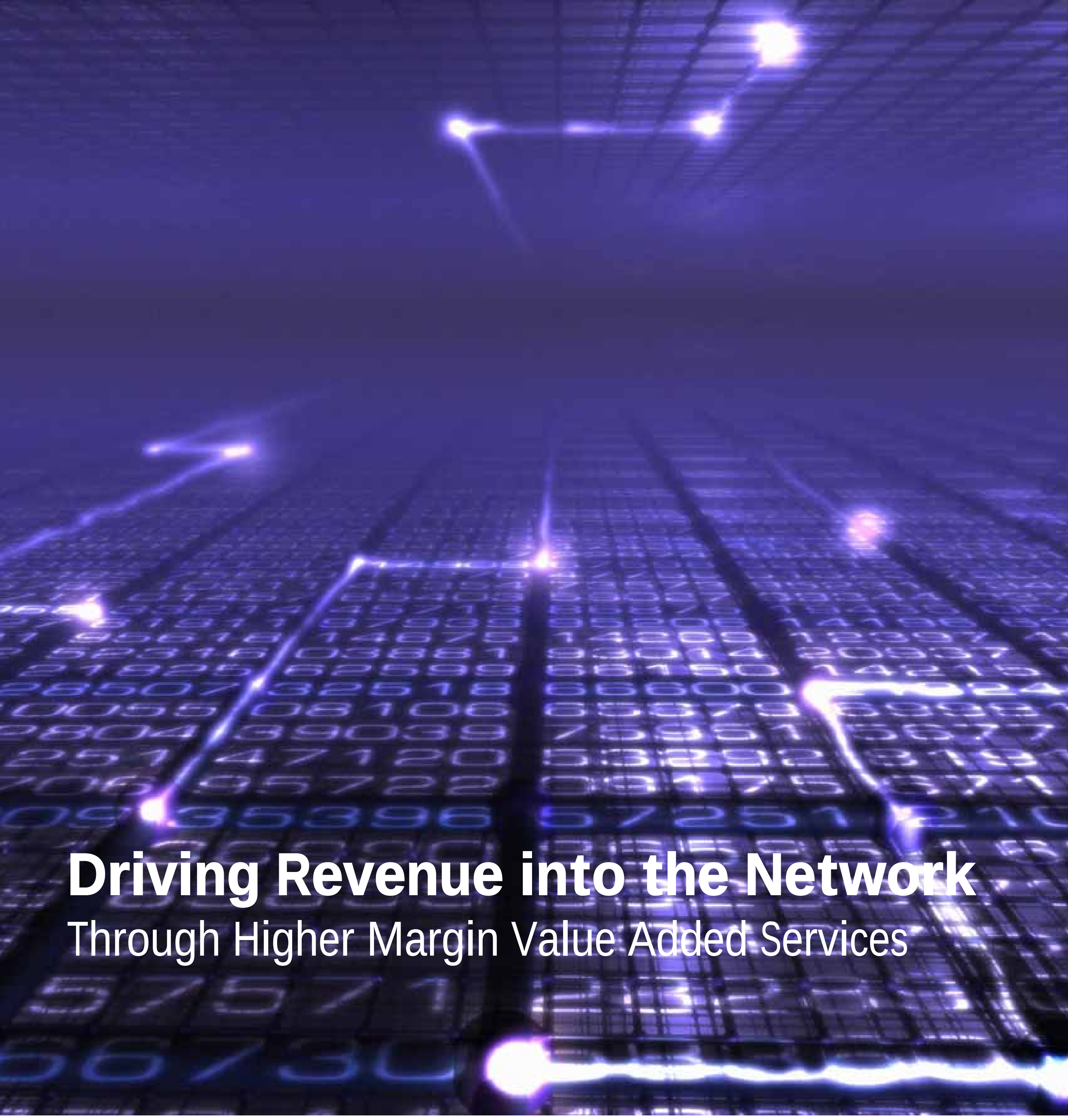

Driving Revenue into the Network

Through Higher Margin Value Added Services

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Background

The telecommunications industry is undergoing a fundamental and profound shift as it migrates from circuit switching to IP based switching. This change ushers in more efficient networks with lower administration and operating overheads, the democratization of the provision of telecommunication services and lower costs to the consumer with far greater functionality and breadth of services.

Juniper Research, a leading research and consulting firm recently stated: "Although much attention has been paid to the more complex and 'sexy' value-added services like video-on-demand and online games, it appears that the biggest opportunity will be in voice services as broadband-IP voice takes over from traditional circuit-switched voice." In the residential market, Juniper reckons online gaming that relies on broadband will rise to being worth $2.9bn globally in 2009, up from $600m in 2004. Music will grow to $2.12bn from $410m, video services to $5.71bn from $800m and voice over IP (VoIP) to $27.1bn from $4.29bn. Business use will see online data storage move to $2.92bn from $310m in 2004 in 2009, Web hosting to $5.28bn from $2.67bn, virtual private network provision to $2.36bn from $550m and VoIP to $20.09bn from $4.54bn. These figures put the combined VoIP market at $47.19bn.

Along with this great opportunity will come increased competition and reduced margins as new companies try to gain market share from the incumbents by competing solely on price. These downward price pressures coupled with growing global deregulation will serve to commoditize long distance services which will be great for the consumer and will serve to increase overall market demand.

New and established service providers are making the investment in their networks today to be in a better position able to withstand the forthcoming market forces by first introducing the more robust and application rich VoIP technology, and secondly, by offering services above and beyond basic call termination services that have higher margin and less competition. These additional services include conferencing, voicemail, automated attendant services and PSTN inbound origination. These services are built around increased customer loyalty with a broader range of integrated services that carry with them higher margins. By combining additional services with feature rich IVR (voice guided prompting and DTMF response) and flexible real-time billing facilities, the service provider has the necessary infrastructure to build highly competitive valued added service offerings that allow for creative marketing to attract and retain customers. With the substantial market growth that industry research firms have forecasted, it is through value added services that telephony providers will be able to drive higher profitability into their network by deploying more efficient network architectures and in-demand and easily adaptable service applications.

Network Solution

The Session Initiation Protocol (SIP) makes the integration of third party applications possible, bringing vendors from all areas of expertise together to form a broadly interoperable, seamless and feature rich environment for service providers to build their value and product offerings from.

IVR Technologies, Inc. and ABP Technology have partnered to seamlessly integrate their best in class products into a single, cohesive revenue generating solution that is remarkably easy to deploy, manage and maintain. We have combined the IVR Technologies' Talking SIP™ IVR and real-time billing solution with ABP Technologies' snom 4S SIP Proxy & Registrar, Media Server and NAT Filter into a complete turnkey solution. The snom 4S Proxy & Registrar provides SIP end-point aggregation, registration and management as well as first tier authentication. Once the subscriber has been authenticated by the proxy they are directed to the Talking SIP™ platform for further authentication (ANI, DNIS, IP Address, Account Number, Account Number + PIN), IVR, rating and routing. Talking SIP™ keeps track of the call state and enforces the billing plan and services that the customer has subscribed to. At this stage the caller can be directed to the appropriate service offering automatically based on their account configuration setting or by voice guided prompting such as "*To access the voicemail server please press 1, for conferencing services please press 2,…".* The snom 4S Media Server then provides the value added service applications in a readily extensible VoiceXML framework. The snom 4S NAT Filter makes it possible for subscribers residing behind a firewall to be fully serviced without encountering any transversal issues as they cross the private and public boundaries without requiring changes to network or device, greatly reducing the overhead for provisioning and managing the customer base.

On the residential or corporate customer side there are multiple types of network access that are possible with this solution, such as the standard or mobile telephone, and if broadband Internet is available devices such as Internet Access Devices (IADs), softphones and IP phones may be used. Once the caller is authenticated into the network, a slew of services such as voicemail, follow me, calling card, travel card and conferencing services are available. Mobile customers have the flexibility of being able to have continuity of service wherever they may be, by having their access device register and update its information while they are on the road without interruption of service. Secure customer self management via the web provides centralized management for automated subscriber signup, account recharge, call detail record tracking, speed dial management and so much more.

Network Components

The following are the network components that are brought together for this seamlessly integrated and cohesive network solution:

Talking SIP™

Talking SIP™ is an advanced software solution for voice applications and real-time billing over SIP networks. Talking SIP™ speaks SIP directly to Industry leading softswitches and gateways to provide applications such as calling cards, tandem, and e-commerce. Talking SIP™ comes standard with integrated real-time prepaid and postpaid billing, reporting and a complete management interface. Built on Windows® 2000/2003 Server and Microsoft SQL 2000 Server, Talking SIP™ runs on off-the-shelf Intel Pentium® processors, so there is no need for costly and special hardware configurations.

snom 4S Registrar Proxy Server

The snom 4S Registrar/Proxy is the main registration and authentication component of the solution. It consists of a location server and registrar according to RFC 3261. The Registrar/Proxy is a scalable solution available in three different versions: ENTRY, SME and EXPERT. The Entry model is intended for customers who wish to set a small VoIP network. The SME version is geared towards the medium-sized company with up to 50 users. The EXPERT version adds features suitable for SIP-based telephony operators with large subscriber numbers.

snom 4S Media Server

The snom 4S SIP Media Server is a turnkey solution for handling media in SIP environments. Using the SIP protocol, the snom 4S Media Server can be used with software solutions such as Microsoft Messenger or with a variety of hard phones such as the snom phone products. The snom 4S SIP Media Server can be used with the snom 4S SIP Registrar/Proxy as well as with other SIP proxies, registrars and location servers. It is able to play and record audio, to react on user input and to control calls. Its behavior is controlled with XML files, making it a flexible solution for a wide range of requirements. Full support for Interactive Voice Response (IVR) systems is built in.

snom 4S NAT Filter

The snom 4S NAT Filter enables non-NAT aware devices to operate in private networks and behind firewalls. The NAT filter operates on a public IP address and is addressed as an outbound proxy. Non-NAT aware devices are automatically refreshed; NAT-aware devices that operate behind symmetrical NAT may self-refresh their bindings using the built-in STUN server of the filter. Devices on public Internet traverse the filter without changes or refreshes.

Network Diagram

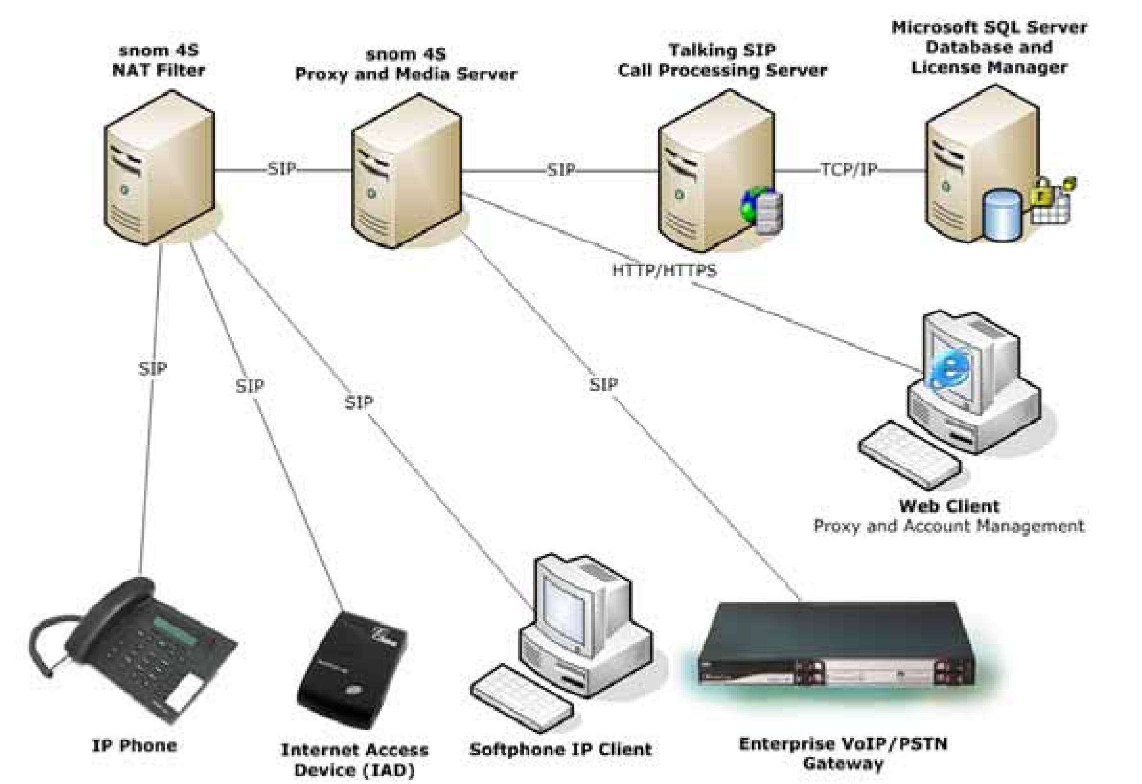

Figure 1. This diagram shows a typical diagram of the network elements of this solution.

Value Added Services

The integrated solution offers many in-demand and extensible enhanced services to the network in prepaid and post paid billing models with a fully integrated and customizable invoicing engine. The following is a brief description of some of the services offered by this solution:

800/900 Termination Module

The 800/888/877/900/DID Termination module allows the termination of toll or toll-free calls to a local phone number and/or SIP device for inbound PSTN termination. This module allows prompt and efficient redirection of toll-free numbers to various termination numbers without the traditional delay and overhead. The 800 Termination module also provides full billing and accounting of all traffic that passes through each toll-free or DID number.

Auto Attendant and IVR

The auto attendant welcomes callers with a customized message and redirects calls to predefined destinations or to extensions that can be entered with DTMF. The auto attendant is controlled by a dial plan that lists the available destinations in a table-like fashion. Wildcards make it easy to set up the auto attendant for large installations. IVR trees can be used to guide callers through voice prompts. Callers can navigate using their DTMF keys. This feature makes it easy to set up large corporation-style central numbers without any programming or XML.

Conferencing

The conference server can mix multiple conferences with multiple participants. It is possible to set up password protected conference accounts. The conference server is able to mix conferences with mixed codecs. The media server supports ad-hoc conferences with or without password.

Debit Card Module

The Debit Card module provides calling card functionality to Talking SIP™. Prepaid and postpaid accounts are authenticated and authorized manually by DTMF key presses or automatically based on the caller's ANI or IP address and then prompted for the desired destination. The caller's destination is authorized and rated and Talking SIP™ then connects the caller through to the desired destination connecting the audio directly between SIP end points while maintaining control of all of the signaling. The call is tracked to detect when the caller wishes to make another call without having to be re-authenticated with maximum call durations enforced by Talking SIP™. When the account's balance is depleted the call is automatically disconnected and accounting records written in the Microsoft SQL database server so the caller's balance can be updated and a call detail record written.

Tandem Module

Tandem Switching allows Talking SIP™ to provide class 4-tandem switching where the platform answers calls and authenticates and authorizes the call and then rapidly terminates the call through the platform. Account authorization can be performed based on a specified account, DNIS, ANI or IP address.

Voicemail

The mailbox subsystem is able to welcome callers with different standard and customized messages that can be set up by the users of the mailbox. Upon reception of new messages, the mailbox sends message waiting indications (MWI) to predefined destinations or to user agents that have subscribed to these notifications. New messages can be forwarded to POP3 email accounts. The recorded messages may be sent to the users via Email. The attachments are compressed via a low-rate speech codec and can be played on email programs like Microsoft Outlook.

Voucher Recharge Module

The voucher module allows service providers and telecom operators the option of providing their customers with the flexibility to transfer balances from one account/voucher to another account. This module can be configured as a stand-alone application or integrated with the Debit Card Module whereby the caller is provided with the option of transferring a balance whenever he/she logs into the system. This option helps to increase customer retention by providing customers with the option of being able to transfer balances to a familiar account number as opposed to having to memorize a new account number and/or reprogram telephone speed dials whenever they wish to extend their credit.

Add On Products (Options)

The following is a list of optional add-ons to the Talking SIP™ product that further extend and enhance the solution within the network:

- **End User Web Interface**
  The End-user web interface allows providers to give their customers the convenience of empowerment of being able to access account information, call detail records, and billing history whenever they desire. Used in conjunction with our Online Credit Card Recharge module, the Web Based Interface gives customers the ability to offer an e-commerce distribution channel for self-service

account provisioning and account recharge to extend the reach of an operation without consuming the costly resources of customer services representatives.

- **Intelligent E-mail Agent**
  The Intelligent E-mail Agent allows system reports such as traffic analysis, call summaries or billing records to be extracted from Talking SIP™ and automatically e-mailed at predefined intervals. System administrators can also use this agent to execute stored procedures or custom SQL statements to assist in the maintenance of the network and/or database. All tasks and reports can be scheduled on a daily, weekly, monthly or an explicit date for a given period of time.

- **Online and DTMF Based Credit Card Recharge**
  The Online Credit Card Module allows account balances to be monitored for prepaid debit cardholders. After an account passes a certain threshold the caller is provided with the choice of having their account recharged online and in real time via a credit card or via telephone and DTMF key presses. This module offers great convenience to the cardholder and generates revenue on the platform without requiring costly human intervention.

Future Extensibility

Experienced users can extend the functionality of the media server by recording new WAV audio files and by setting up their own account types in XML files. In this way, functions like wake-up services and calling card can be implemented. Using the media server together with a forking proxy like the snom 4S Registrar/Proxy, call distribution can be implemented. The media server can be used in a server farm for maximum scalability. It supports DNS NAPTR (ENUM), SRV, loose routing and TCP transport layer for maximum compatibility with other SIP components.

The Media Server can be scripted like a state machine. Using a simple XML-based language, transitions can be described based on state and events. This allows for very flexible application development.

Talking SIP™'s external telephony loadable modules may also be extended and customized to meet and respond to changing future market demands by either acquiring a license to this Basic/Pascal type scripting language or by allowing IVR Technologies' Professional Services group to implement the customizations, for an additional fee.

Network Management

With limited administrative resources, the ability to easily deploy, configure, manage and support this multi-component solution is vital to a next-generation service provider's success. All of the network elements can be managed centrally or remotely either via a common Web browser, or a the freely distributable management console.

Figure 2.

Talking SIP™ is managed through the Telephony Management Console™ (TMC) that allows one or more communications nodes to be centrally managed in a Microsoft Management Console™/Microsoft Outlook™ styled interface. The interface has extensive online and context sensitive help to guide the administrator and/or customer service representatives in their operation of the platform. A robust and tiered security and logging model ensures that access is provided on an as-needed basis with a full audit trail.

snom 4S Registrar/Proxy Server

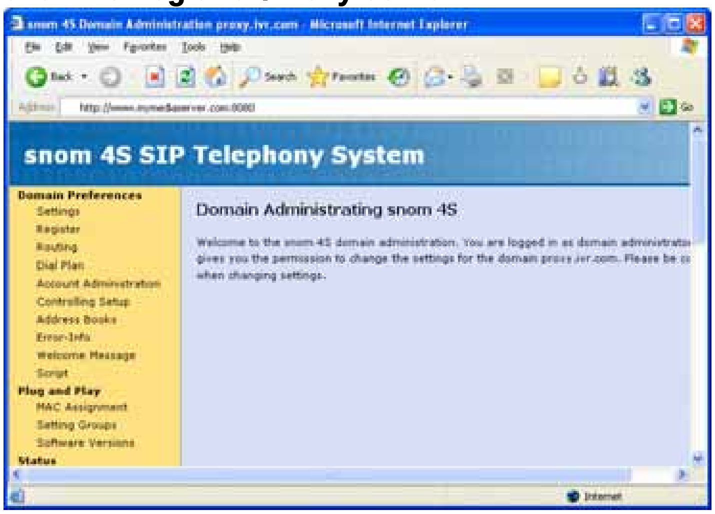

Figure 3.

The snom 4S Registrar/Proxy server is managed via its built in web server to provide ubiquitous and full access for Administrators and End Users. This secure interface provides Administrators access to all of the system functionality for domain/account creation, ongoing maintenance and management as well as troubleshooting. End users can also use this interface to manage their accounts for such features as call forwarding, call routing, no answer settings and to review their call history.

snom 4S Media Server

Figure 4.

The snom 4S Registrar/Proxy server is managed via its built in web server to provide ubiquitous and full access to Administrators and users. This secure interface provides access to all of the system functionality for deploying services, managing codecs and configuring network settings.

End User Web Interface (Optionally)

Figure 5.

Talking SIP™ provides an optional end user web interface to provide end customers the ability to self subscribe for services and well as the flexibility to manage their account at their convenience for account recharge, updating contact details, setting speed dials, updating their password or for reviewing account activity.

Opportunity

CNET News reports that the number of U.S. Internet telephony subscribers is expected to grow to 1 million by the end of this year from 131,000 in 2003, according to a new study. This number is expected to surge to 17.5 million households by 2008, according to the study, released by The Yankee Group.

This major prediction of subscriber growth for Internet based telecommunications provides a significant market opportunity for next-generation service providers to attract and retain subscribers by launching differentiating services. The virtual and global design of SIP makes it possible to market these services to both domestic and International markets, thus not limiting the service provider to any one specific region.

IVR Technologies, Inc. and ABP Technology are dedicated to bringing high value solutions and expertise to SIP based next-generation service providers. Call us today for more information about this exciting emerging market.

Contact Us

Please visit us on the web at (www.ivr.com or www.abptech.com) or contact us directly to find out more information about our products, receive a quotation or locate a value added reseller in your region:

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