Telecoms related products and services from Muftasoft ™

T1,E1 and PRI Digital VoIP Gateways

A **VoIP PRI Gateway** enables direct routing of calls between the fixed line ISDN and the cost-effective IP networks for capitalizing on low cost VoIP telephony. By integrating a VoIP PRI Gateway with existing PBX / PABX telephone systems, businesses of all sizes can benefit from low-cost Voice over IP calls and achieve substantial cost savings without the high upfront costs associated with communications infrastructure changes related with the migration to VoIP .

Audio and Video Conferencing

Organizations are realizing the benefit that face-to-face online interaction brings to their business operations. In response, they are looking for conferencing solutions to help them meet their needs while staying within budget. With digital conferencing increasing in popularity, IT professionals are commonly asked: What is the difference between audio, video, and web conferencing? There are defining aspects and advantages to each and knowing the ins and outs of the different formats can help you decide which solution will work best for your business.

Web Conferencing

Web conferencing involves sharing content live over the web between two or more locations. Visual material is usually accompanied by an audio track. You can use web conferencing to do live presentations, team meetings, or employee training. Typically, participants connect from their own devices to the conference via the Internet. There are different types of web conferences, but the two most commonly used are webinars and webcasts. A webinar is a one-way conference between one or more speakers. A webcast is a live streaming event, broadcast over the Internet.

Video Conferencing

Video conferencing has been widely adopted and is fast becoming the most common form of digital collaboration. Two people having a chat? You're good to go. But when more than two people are joining a video conference, you will require a bridge device, usually a multipoint control unit, often found within a company network. However, they are also available through a third-party provider. There is commonly a monthly subscription based fee with providers. But, some companies supply the service for free. This technology is gaining popularity because visual elements enrich conferencing. Additionally, it lets organizations conduct meetings or pitches remotely, greatly reducing travel costs. Video conferencing also offers audio-only capabilities, file sharing, and screen sharing options. Calls are made on the company's internal network, but if an outside party needs to join in, security usually dictates doing so over the Internet behind a firewall. To make video conferencing most effective, it's important to have good lighting and stand/sit directly in front of the camera. This way, participants appear to be in front of one another, giving users the feeling of a face-to-face meeting, despite being in different locations. Some trends and issues that are emerging in the video conferencing space include: ***Web real-time collaboration (WebRTC)***. This allows the use of instant messaging, chat, presence, ad hoc audio, and document sharing. ***Mobility.*** Video conferencing applications that support mobile are increasingly important. The key challenges are quality issues with WLAN connections and battery life. ***Cloud solutions.*** These are efficient solutions as they require little upfront cost and IT knowledge. However, there are concerns in regards to keeping up to date with current software and incompatible technologies.

Audio Conferencing

Audio conferencing has been around the longest and this method allows many people to participate in a call by connecting through a conferencing bridge. With an ID and access number or unique code, you can log into a group call. This may also be supplemented by individual PIN numbers to identify callers and/or for security reasons. If someone wants to share documents, set an agenda, or provide background information, they usually will need to do so externally in advance of the call, via email, for example. Audio conferencing is slowly losing favor due to the popularity of video conferencing's more personal, face-to-face interaction, though there are still some who prefer the more traditional method. Remember to invest in a high-quality microphone to get the most out of your audio conferencing.

Cloud vs. Desktop Solutions

The cloud now offers a viable alternative to desktop solutions. The main benefits of cloud video conferencing are scalability and cost reductions. With many employees now working remotely, cloud videoconferencing can allow for more flexible work hours, and cut travel costs for your firm. Saving money is obviously an incentive, but time saving is even more important. The main things to consider with cloud solutions are: â¯¢ Learning how to effectively manage a remote workforce without spending a lot of money. â¯¢ Learning how to employ cloud video conferencing without having to invest heavily in IT. â¯¢ Keeping your video conferencing software current and being prepared for the future. â¯¢ Encouraging your employees to adopt the technology. â¯¢ Ensuring your cloud solution's compatibility across all your devices and applications. Web, video, and audio conferencing each provide their own benefits. With the technologies becoming more user-friendly and cost effective, businesses now have more choices open to them. Organizations can choose one, like video conferencing, to fit all their needs or adopt more than one, to be used for different formats or occasions. By knowing the differences and what each conferencing type entails, a company can choose the method(s) that will be the best for their business.

IP Conference Phones for multi location and Multi party IP conference

Make it feel like everyone's together in the same room. Conference phones are useful because they deliver the clearest sound to every participant in every location. The advanced audio technology allows each conference phone to intelligently adapt to different room environments. So everyone can hear and be heard, even when more than one person talks at a time. You'll eliminate confusion and enhance productivity. Not a single word—or opportunity—gets missed.

Headsets

A headset combines a headphone with a microphone. Headsets are made with either a single-earpiece (mono) or a double-earpiece (mono to both ears or stereo). Headsets provide the equivalent functionality of a telephone handset but with hands-free operation. They have many uses including in call centers and other telephone-intensive jobs and for anybody wishing to have both hands free during a telephone conversation.

Skype For Business

Skype for Business, formerly known as Microsoft Lync Server, is a unified communications (UC) platform that integrates common channels of business communication and online meetings, including instant messaging (IM), presence, voice over IP (VoIP), voicemail, file transfers, video conferencing, web conferencing and email.

Asterisk Based VoIP/SIP/IAX2 servers.

Asterisk is a complete PBX in software. It runs on Linux, BSD, Windows (emulated) and OS X and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in four protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.

Asterisk provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response, Call Queuing. It has support for three-way calling, caller ID services, ADSI, IAX, SIP, H.323 (as both client and gateway), MGCP (call manager only) and SCCP/Skinny. Check the Features section for a more complete list.

Asterisk **needs no additional hardware for Voice-over-IP**, although it does expect a non-standard driver that implements dummy hardware as a non-portable timing mechanism (for certain applications such as conferencing). A single (or multiple) VOIP provider(s) can be used for outgoing and/or incoming calls (outgoing and incoming calls can be handled through entirely different VOIP and/or telco providers)

For interconnection with digital and analog telephony equipment, Asterisk supports a number of hardware devices, most notably all of the hardware manufactured by Asterisk's sponsor, Digium. Digium has single and quad span T1 and E1 interfaces for interconnection to PRI lines and channel banks . In addition, single to quad port analog FXO and FXS cards are available and are popular for small installations. Other vendors' cards can be used for BRI (ISDN2) or quad- and octo- port BRI based upon CAPI compatible cards or HFC chipset cards.

For interconnection with the cellular network (GSM or CDMA), Asterisk can use the Celliax channel driver or chan_mobile that is in the trunk now and there is also a unofficial backported version.

Lastly, standalone devices are available to do a wide range of tasks including providing fxo and fxs ports that simply plug into the LAN and register to Asterisk as an available device.

VoIP Adapters

FXO and FXS Gateways

Many businesses today are implementing Voice over Internet Protocol (VoIP) phone systems in their office, to replace traditional phone lines – and for good reason. VoIP provides significant advantages, including the ability to make and receive calls from anywhere and at any time, cost savings, a more reliable signal, and more. But before you start implementing VoIP, you need to understand the difference between FXS and FXO ports. We'll help you get started.

FXS and FXO are the interfaces for analog telephony, also called POTS (Plain Old Telephone Service). Using these interfaces enables a call to be established – the ports provide the necessary electricity, dial tone, and call signal.

What is an FXS port?

FXS stands for Foreign Exchange Subscriber, a port that connects the router or access server to end-user equipment such as office phones, fax machines, or modems. In other words, it is a plug on the wall that delivers dial battery, loop current, and ringing voltage to the device, so that the analog signal can be transmitted.

What is an FXO port?

FXO stands for Foreign Exchange Office, a port on the end communication device, such as an office phone or fax machine. The FXO connects the device to the FXS port, as well as to the outside telephone line, requesting the dial tone needed to initiate a call.

How the connection works

In order for a call to work, a telecommunication line from an FXO port must be connected to an FXS port, and vice versa. The process for making calls is simple: when your FXS and FXO ports are connected, you will receive a signal from the telephone company through the FXS port in the wall. This signal is then transmitted to the FXO port connected to the device so that, when you pick up the phone, you hear the dial tone. Then you dial the phone number, which is passed as Dual-Tone Multi-Frequency (DTMF) digits to the FXS port, allowing you to make the call.

When you're receiving an inbound call, on the other hand, the FXS port receives the call, then sends a ring voltage through the FXO port to your end device. The phone will ring, and you can pick it up to answer the call.

FXS, FXO and VoIP

The signal transmission process becomes more complex when you implement an additional network element, such as VoIP gateway. You need an FXS gateway to connect one or more lines to a VoIP system or provider. You'll also need an FXO gateway to connect the VoIP system with analog phone lines, and to translate the analog phone line to a VoIP call.

In summary…

- FXS is a plug on the wall, and FXO is a plug on the phone
- FXS provides the dial tone, and FXO requests it
- FXS is a port that receives a call; FXO is a port that initiates it

Our FXS gateways include gateways with a various number of ports and support SIP, H.323, IAX2 and MGCP protocols.

We know that deploying a new VoIP phone system doesn't always mean that you need new VoIP phones to go along with it. Connecting to VoIP services via an FXS Gateway is a simple and cost effective way to wrestle extended returns from your existing analog phones or fax machine.

FXS Gateways employ RJ11 ports that will leverage your analog office phones and fax machines to communicate with your new VoIP PBX system with VoIP protocols and will encode/decode the voice signal with voice codecs.

FXO Gateways connect your devices to an outside telephone line and support Failover or Fallback in the event of internet failure or a network crash. An FXO gateway can be implemented to provide access to multiple POTS lines; the gateways normally come in 1, 2, 4, and 8-port configurations.

Supported protocols include SIP, PSTN, H.323 and ISDN.

For businesses with multiple offices, FXO Gateways keep local calls local. When connected to a centralized IP PBX at the main office, FXO gateways can utilize the local landlines of your remote, out of state locations thereby eliminating long distance charges.

Smart & Managed Ethernet Switches

Unmanaged switches have a set of basic features but no options for configurations. They *just work they way they do* ~ plug&play.

Managed Switch Vs. Unmanaged Switch

Managed switches, mainly features like VLANs, Port Mirroring, SNMP, adjusting the port speed

Most managed switches offer you features like:

---View the bridging table to see which MAC addresses are associated with a given port
---View error statistics for each port
---View packet transmit / receive statistics for each port
---Set duplex / speed negotiation (or lack thereof) on a per-port basis
---View power-over-Ethernet status and current draw for each port (if applicable)
---support things like 802.1D spanning tree, 802.1q VLANs, and 802.3ad link aggregation,
---setup port VLAN memberships, link aggregation groups, and control spanning tree parameters all from a web or command-line interface
Typically there is a TELNET, serial, and / or web-based interface to interact with the switch.Many managed switches allow you to poll the device with the SNMP protocol to use the information described above in graphs, alerts, etc..Some managed switches are to emulating the Cisco command-line interface (HP ProCurve, Dell PowerConnect....) such that someone with Cisco-specific knowledge can easily configure those switches.

Routers

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