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INTERNET2®

AASTRA®

Your Voice in the Cloud

Internet2 NET+ SIP Services

Internet2 NET+ SIP Services

Universities don't need to be their own phone companies any longer—and don't need to spend millions of dollars on capital expenses. Internet2 is partnering with two voice services industry leaders, Aastra and Level 3 Communications, to offer a cost-effective alternative to traditional voice services.

Aastra offers a hosted PBX service that includes traditional PBX features, unified communications, and call center services. Level 3 offers SIP-based services such as local trunks, long-distance trunks, toll-free and other services.

These services are based on the industry standard SIP protocol and designed to provide flexibility, local control and local survivability, and empower institutions to customize the solution to meet their unique campus needs.

Complete Cloud Communications Solution

HOSTED PBX

- Extensive range of PBX and UC features
- Fixed-Mobile Convergence
- Desktop telephony integration
- Unified Messaging
- Call Center
- Auto Attendant
- Receptionist

SIP TRUNKING

- SIP Trunks
- Flat rate trunks with long distance
- Metered long distance
- Toll free Services
- Number portability
- E911

CARRIER-CLASS SOLUTION

- Geographical redundancy
- Site survivability
- Based on open standards
- Additional options with testing supported by ITEC
- Leverages Internet2 backbone

Benefits of Hosted Services

Lower TCO
Hosted solutions distribute costs over many customers, including costs for equipment, software, facilities and management. With Internet2 SIP Services, the Total Cost of Ownership (TCO) over a 5-year period is lower than competing hosted service offerings, and either less than or comparable to a premise-based deployment. And, it gets even better. As the total number of users for the entire SIP Services offering increases, the price decreases for all SIP Services subscribers.

Better utilization of resources
A hosted communications solution allows you to redirect your IT resources to projects that are more strategic to your organization. Unified communications is not a core competency for most institutions. Just as you may already outsource things such as email, storage, or various administrative applications, why not move your voice communications to the cloud and use your IT resources on projects to accomplish your strategic goals and give your organization a competitive edge?

Business continuity
Automatic disaster recovery is a key part of NET+ SIP Services, and is much less expensive than doing it on-premise in your own data centers. In addition to system component redundancy, the SIP Services offering is designed with geographical redundancy, and also offers site level survivability.

Lower risk of obsolesce
Premise-based solutions may offer the latest features and functions when they are first deployed, but they can become quickly obsolete. With SIP Services you get a state-of-the-art solution which will remain up-to-date. Software upgrades, patches and other updates of all the components in the solution are routinely applied for you at no additional cost.

Reduced complexity
Hosted solutions for unified communications are easier and faster to deploy. You don't have to deal with multiple vendors and complex interoperability issues. With SIP Services you get a fully integrated solution, and you can be up and running in a matter of days, not months.

Benefits Unique to NET+ SIP Services

In addition to the benefits of using hosted services, the NET+ SIP Services offers many unique benefits which may be not available with competing solutions.

Lowest pricing
By leveraging the combined buying power of the members of the community, Internet2 was able to negotiate rock bottom pricing. A complete package for an end user including hosted PBX, SIP trunking and long distance services is around $6.00 per month per user. We challenge you to find a better price anywhere!

Open standards solution
The services are built completely on open standards including the SIP protocol. This gives schools options as far as the endpoints they would like to use and other applications they would like to integrate.

Backed by the ITEC lab
The Internet Technology Evaluation Center at Texas A&M University runs a system identical to the platform powering the SIP Services offering. ITEC has fully tested the solution and can provide additional testing for new services, additional SIP endpoints and applications that universities may want to use. For more information visit http://itec.tamu.edu.

Local control
One concern with hosted services is that institutions worry that they will lose control of adding or changing users, or customizing features. With SIP Services this is not the case. For example, the hosted services solution allows local administrators to perform any moves, adds, changes and to customize what features users have access to. This control can be further pushed down to various departments or groups if the administrator chooses to allow it. And, individual users can easily configure their own features such as find-me/follow-me, selective call acceptance or speed dial numbers.

Webinar
Want to learn more? View the webinar recording with Adobe Connect.
Click here to get started »

A complete package for an end user including hosted PBX, SIP trunking and long distance services is around $6.00 per month per user. We challenge you to find a better price anywhere!

Hosted PBX Services are offered using Aastra's Clearspan® platform.

Clearspan is a complete solution for unified communications providing all of the applications your users need, at any of the locations where they work, on any of the devices they use. It not only offers classic PBX-like services, but also a rich set of enhanced services including unified messaging, fixed-mobile convergence, interactive voice response, conferencing, collaboration and call center applications. Clearspan allows employees to make and receive calls from any device, at any location, with only one phone number, one dial plan, one voice mailbox, and a unified set of features.

Clearspan Assistant is an integrated toolbar that enables users to make and accept telephone calls, change telephone settings and access directories from within a web browser. In addition, the Clearspan Web Portal allows users to configure personal call services including Call Forwarding, Anonymous Call Rejection, Do Not Disturb, Simultaneous Ringing, Remote Office and Voicemail.

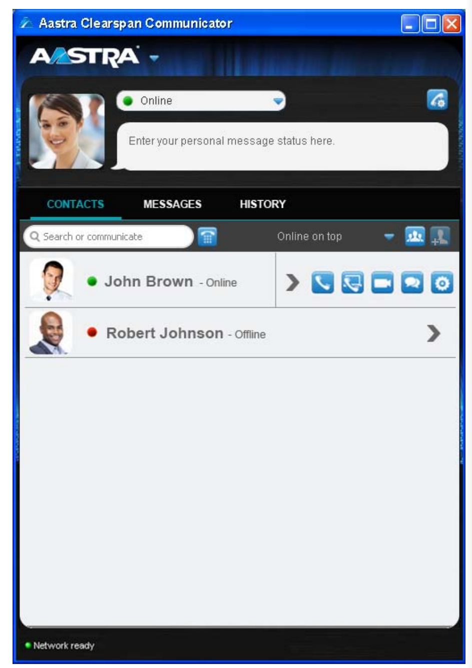
Clearspan Anywhere™ is a native fixed-mobile convergence solution within Clearspan that lets end users designate a single phone number for all incoming and outgoing calls, regardless of which phone they are currently using — mobile, office, home, etc. Calls to a single number ring all phones and the user is free to answer on any device. Once a call is active on the single number, the user is free to pick up any other device and continue the conversation uninterrupted. For example, you can easily move a call from your office phone to your mobile (or vice-versa) without interrupting the conversation.

Clearspan Unified Messaging goes beyond classic voicemail and provides users with the flexibility to use and manage their messaging service from anywhere, and over any interface. Clearspan messaging provides all of the features of a traditional voice messaging solution, as well as message delivery to any specified email account, fax messaging, and much more.

Clearspan Receptionist is an integrated, desktop client application that provides an IP telephony attendant console for use by enterprise receptionists. The objective of Clearspan Receptionist is to simplify business processes and deliver critical information in real time to accommodate professional call handling.

Clearspan Call Center provides a range of options from Standard to Premium and clients for Agent or for a Supervisor. Whether your institution needs basic ACD functionality for a single workgroup or a solution for a large number of agents spanning multiple locations, Clearspan's Call Center solution can be tailored to fit your needs.

Clearspan Communicator™ is a unified communications client for the PC, MAC, iPad, iPhone and Android platforms. Clearspan Communicator can be used to make and receive both voice and video calls as well as initiating call control features. Clearspan Communicator integrates with the enterprise directory and personal contacts, includes call logs and can be used to manage features such as Clearspan Anywhere™.

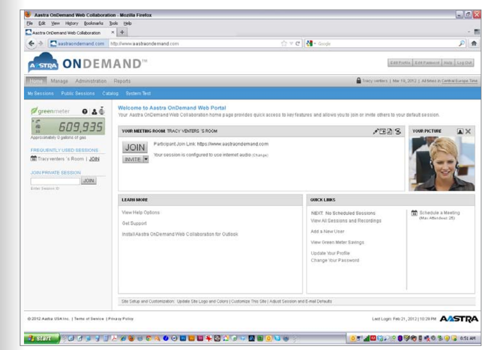


Clearspan Communicator™

Clearspan Assistant Toolbar

Hosted PBX

Aastra OnDemand™ Web Collaboration is a feature-rich web and video conferencing tool for hosting online meetings, webinars and training. Aastra OnDemand provides users with the ability to meet and collaborate online including voice and video conferencing, sharing documents and applications, interactive whiteboarding, group or private chat, polling and much more.



Aastra OnDemand Web Portal

Meet-Me Conferencing provides users with an audio conferencing service supporting both reservation-less and scheduled conferences. It is a full featured platform including the ability to mute/unmute participants, lock/unlock the conference, outdial to participants, and record conferences. It also includes a web-based moderator client as well as integration with Microsoft® Outlook™ for notifications to participants.

Additional Hosted PBX Resources

[Clearspan® Brochure (PDF) »]

[Hosted PBX Feature List (PDF) »]

[SIP Phone Choices (PDF) »]

Aastra SIP Phones

Built from the ground up with Session Initiation Protocol (SIP) at its core, Clearspan supports a wide range of endpoints. This includes desk phones from Aastra or 3rd parties, soft phones, mobile and video clients. And since it provides carrier-grade reliability, it is the perfect choice for institutions that require high-capacity, mission critical performance.

While all Aastra SIP phones come with a one year warranty, institutions can optionally purchase an enhanced warranty to extend the total warranty period to 3 or 5 years. The enhanced warranties include advanced replacement of failed units.

6731i *6735i / 6755i*

600d DECT handsets *6739i*

Hosted Services Video

Want to learn more? View the Internet2, Aastra Hosted Services video. Click the video icon to the left or click here to get started »

AASTRA®

Frequently Asked Questions

Do I have to be a member of Internet2 to take advantage of these services?
The primary condition is that your organization must be connected to the Internet2 network. If your organization is not a direct member of Internet2, then a more detailed discussion needs to happen. Not sure if you are member? Visit http://members.internet2.edu/university/universities.cfm to find out.

If I am not an Internet2 member how can I become one?
Visit www.internet2.edu/membership to learn more about membership and the unique, advanced technologies Internet2 provides to the U.S. Research and Education communities or contact membership@internet2.edu for more information about becoming a member.

What are the prices for the services?
There is a complete pricing sheet available for all of the base services and optional services. Working with several schools, Internet2 developed a general estimate of an average cost per line of approximately $6 per month. Each school will pay the actual costs for the specific services and features they purchase. As the Internet2 members collectively increase the number of lines in service, the costs for all the schools will be reduced.

Where is there an overview of all the Hosted PBX features available?
The details of the SIP Services are available at the link below.
Hosted PBX Feature List »

How does the SIP phone service integrate with Microsoft® OCS (or Lync)?
Aastra's Clearspan service has the capability to integrate with Microsoft OCS/Lync. There are many options and Aastra would work directly with the members to customize a solution. Level 3 SIP Trunking fully supports Microsoft Lync and is the SIP and Infrastructure provider for the Microsoft Lync Reference Architecture project.

Is there a contractual commitment for the service? If so, what are the terms and conditions?
Yes. The service is designed on a five year program to make sure the schools and the service providers don't invest a lot of energy in migration without a guarantee of reliable service over an extended period. Terms and conditions will be covered under the services agreement. Contact sip@internet2.edu to find out more.

What telephone numbers can I/will I use?
If you choose to use Aastra's Clearspan as your dial tone replacement but keep your current provider for trunks, porting is not required. If you choose to use Level 3 for trunk services, Level 3 will assist in porting your numbers from your current provider to Level 3.

Can you describe the migration process to a SIP dial tone service?
For SIP phone service, there are generally two options: 1) Add new features to a legacy PBX or 2) Full VoIP deployment from Internet2's SIP Services on the Aastra Clearspan service. If option 1 is selected, SIP trunks are put in place between your existing PBX and the Aastra Clearspan service. Routing between the two systems will need to be set up and tested. If option 2 (preferred) is chosen, numbers/extensions are assigned, phones are shipped and deployed and users can begin using the services immediately. Hybrid scenarios (a combination of both options) are also available.

How long will it take to migrate from my current system to this new service?
Service turn-up, from Aastra, typically requires 45-90 days. Full campus migrations may be spread out over a 12 month period. If you choose Level 3 for trunk services, Level 3 generally recommends 6-8 weeks once a plan has been devised. If you choose both services, both Aastra and Level 3 will work with your organization to devise a comprehensive migration strategy.

What if the Internet2 network is down? Will I lose my ability to make and receive phone calls?
Local survivability options are available from both Aastra and Level 3. There will be standard survivability configurations available or one can be customized to your organization's requirements.

For more information visit www.internet2.edu/sip or email us at SIP@internet2.edu.

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