Pexip Infinity

Administrator Guide

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The Pexip Infinity platform

Pexip Infinity is a scalable meeting platform that connects virtually any communications tool, such as Microsoft Lync / Skype for Business, and traditional video and audio conferencing together for a seamless meeting experience.

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Introduction to Pexip Infinity

Pexip Infinity is a virtualized and distributed multipoint conferencing platform. It enables scaling of video, voice and data collaboration across organizations, enabling everyone to engage in high definition video, web, and audio conferencing. It can be deployed in an organization's datacenter, or in a private or public cloud such as Microsoft Azure or Amazon Web Services, as well as in any hybrid combination.

It provides any number of users with their own personal Virtual Meeting Rooms, as well as Virtual Auditoriums, which they can use to hold conferences, share presentations, and chat. Participants can join over audio or video from any location using virtually any type of communications tool (such as Microsoft Lync / Skype for Business, a traditional conferencing endpoint, a mobile telephone, or a Pexip Infinity Connect client) for a seamless meeting experience.

Virtual Meeting Rooms and Virtual Auditoriums can also be accessed through a Virtual Reception IVR service, which allows all participants to dial a single number to access Pexip Infinity, and then use the DTMF tones on their endpoint to select the conference they want to join. The platform also includes the Pexip Distributed Gateway service, allowing end users to place calls to other endpoints that use different protocols and media formats.

The complementary Pexip Infinity Connect suite of clients allow users to connect to any Virtual Meeting Room or Virtual Auditorium within the Pexip Infinity deployment, either:

- directly from a web browser without any special downloads or plugins
- from an installable desktop client
- from an Infinity Connect Mobile client, available for iOS or Android.

In addition to connecting with video and audio, Infinity Connect users can control the conference, view presentations, share content and chat. Infinity Connect can also be used to make direct calls to other devices or systems when used in conjunction with the Pexip Distributed Gateway.

For more information on using and administering Infinity Connect, see Introduction to Infinity Connect.

Pexip Infinity's unique distributed architecture is purely software-based and virtualized, running on industry-standard servers, meaning it can be deployed quickly and simply with the flexibility to scale as required.

Pexip Infinity version 14.1 features and specifications

The Pexip Infinity platform is designed to use industry-standard servers from any vendor to provide high-quality, scalable and efficient conferencing. The following tables cover the platform, Infinity Connect, audio and video, host hardware, capacity and hypervisor specifications and requirements.

Pexip Infinity platform

Feature	Description
Application deployment and management	• Software-based, virtualized application architecture, running on industry-standard servers. • Management using industry-standard tools, including VMware vSphere, Microsoft Hyper-V, KVM and Xen, and the ability to deploy onto generic hypervisors and orchestration layers. • Ability to deploy on Amazon Web Services (AWS) and Microsoft Azure cloud platforms, including dynamic bursting into the AWS cloud when primary conferencing capabilities are reaching their capacity limits. • Flexible deployment model allowing customers to deploy the platform in the way that is most appropriate for them without needing to consume additional software licenses or purchase dedicated hardware. • Ability to seamlessly increase capacity by deploying new, updated, or additional hardware resources. • Management API supporting configuration, status reporting and call control. • Support for Russian and Chinese language in the Pexip Infinity Administrator interface.
Distributed architecture	• Efficient distribution to reduce bandwidth consumption over expensive WANs. • Industry-leading resilience and redundancy capabilities. • A flexible licensing model that allows you to pool conference resources and quickly increase capacity in response to current local requirements. • Able to overflow capacity between nodes and locations, providing support for conferences that span multiple physical boxes. • Keeps media as local to each endpoint as possible, reducing the negative impacts of latency, jitter, and packet loss commonly experienced on centralized deployments.
Intelligent conference management	• Upscaling all connected participants to provide a seamless experience to all. • Ability to respond dynamically to fluctuating network conditions by downspeeding and upspeeding individual participants, and support for endpoint-based packet loss recovery and adaptation methodologies (such as packet loss concealment and dynamically adapting bandwidth), thereby protecting the user experience in the event of information loss. • Bandwidth-optimized content sharing towards Infinity Connect clients for crisp image at low bandwidth. • Full support for individual transcoding and transrating of both main stream video and audio, and dual stream content. • Simple conference management and interaction for conference participants using Infinity Connect clients, including the ability for Host participants to add, disconnect, mute and unmute other participants. • Advanced conference management and interaction for administrators (using the web-based Administrator interface or the management API). • Optional tagging of services to allow service providers to track VMR use in CDRs and logs.

Feature	Description

Conferencing services	• **Virtual Meeting Rooms** providing personal meeting spaces for everyone within the organization.
	• **Virtual Auditoriums** designed to hold larger lecture-style conferences.
	• **Virtual Reception** IVR (Interactive Voice Response) service.
	• **Pexip Distributed Gateway** enables endpoints to make point-to-point calls to other endpoints that use different protocols and media formats (e.g. from Lync / Skype for Business or WebRTC to H.323). Includes DTMF support.
	• VMRs and devices can be bulk-provisioned from directory information contained in a Windows Active Directory LDAP server, or any other LDAP-accessible database.
	• Choice of layouts: main speaker only; main speaker + 7 video thumbnails; main speaker + 21 video thumbnails; or 2 main speakers + 21 video thumbnails (see Selecting the layout seen by participants for more information).
	• Conference participants can chat and share messaging content.
	• Can output a dedicated multimedia stream to enterprise CDN (Content Delivery Network) streaming and recording services such as Wowza, Adobe, VBrick, Abiliteam, Qumu and Azure Media Services, and to public streaming services such as YouTube.
	• Ability to manage conferences and participants:
	  ○ PIN-protect conferences and differentiate between Hosts and Guests.
	  ○ Lock a conference to prevent any further participants from joining.
	  ○ Transfer a participant to another conference.
	  ○ Limit the number of participants in a conference, on a per-conference basis.
	  ○ Limit the bandwidth used by each participant, on a per-conference and/or global basis.
	• Ability to re-brand with your own images and voice prompts, on a per-conference basis.
	• Ability to re-brand the Infinity Connect experience.
	• Ability to integrate Infinity Connect (WebRTC/RTMP) functionality with third-party applications and websites via a front-end SDK.
	• Call policy decisions can be taken by an external system or a local policy script.
	• Test call service that allows users to check their connectivity and the quality of their video and audio.

Broad interoperability and protocol support	• Full support for existing industry-standard protocols (SIP, H.323), as well as other technologies (HTML5, Microsoft Lync, Skype for Business, RTMP, WebRTC, ORTC). • Ability to enable and disable support for individual audio and video codecs. • Easy integration with existing SIP and H.323 call control solutions including Cisco UCM, Cisco VCS, Polycom CMA, Polycom DMA, Avaya Aura, Microsoft Lync 2010 and 2013, Skype for Business and others. • Conferencing Nodes can act as SIP registrars and as H.323 gatekeepers. • Support for automatic call escalation using Multiway (Cisco VCS), call transfer capability (Cisco UCM), and CCCP to a Microsoft Lync / Skype for Business meeting. • Support for presence and customizable avatar published to a Microsoft Lync / Skype for Business client. • Support for automatic dial-out to audio bridges, including automatically issuing conference aliases and pass codes via DTMF tone generation. • IPv4 and IPv6 support. • Ability to tag management, call signaling, and media packets independently with DSCP QoS support. • Support for Forward Error Correction (FEC), downspeeding, bandwidth throttling, and other packet loss concealment technologies.
Firewall traversal	• Support for static NAT. • Support for static routes. • Conferencing Nodes can be deployed with dual network interfaces. • Support for far-end NAT traversal (media latching). • Support for media over a TCP connection to assist with firewall traversal.

Feature	Description
Security and monitoring	• Designed to comply with US Federal security requirements. • TLS certificate management, HSTS, certificate signing requests (CSRs). • DTLS support. • Active Directory / LDAP integration for administrator account authentication and authorization. • SNMPv2c and SNMPv3 support. • Support for multiple roles of access. • Authenticated SIP trunks. • Limit Pexip Distributed Gateway calls to registered devices only.

Pexip Infinity Connect

Pexip Infinity Connect is a suite of free client software allowing users to connect to Pexip Infinity services from a web browser, installable desktop client, or mobile device.

Feature	Description
Standard features for all Infinity Connect clients	• Can be used to join conferences as a full audio/video participant, an audio-only participant, or as a presentation and control only participant. • Can be used to make point-to-point calls in conjunction with the Pexip Distributed Gateway. • Provides conference control to Host participants. • Allows participants to share and view content, whether or not they are connected with video and/or audio. Supported formats are JPEG, BMP, PNG, GIF and PDF. • Infinity Connect via Chrome and Infinity Connect desktop client users can share their screen in addition to sharing images and PDFs. • Chat (Instant Messaging) support. • Supports sending of DTMF tones. • Access to this feature can be administratively disabled.
Infinity Connect Web App	• Allows participants to join a Virtual Meeting Room or Virtual Auditorium, or make a call via the Pexip Distributed Gateway, using a web browser as their video endpoint. Supported in: • Google Chrome version 43 and later • Mozilla Firefox version 39 and later • Opera version 23 and later • Microsoft Internet Explorer version 10 and later (requires Flash Player 11 and later ActiveX® plug-in, and must not be in Compatibility View) — note that support for Internet Explorer on Windows 10 systems has been deprecated • Microsoft Edge version 20.10532 or later for WebRTC support (earlier versions will connect over RTMP and use Flash video) • Apple Safari version 6 and later (Mac OS X only) (requires Flash Player 11 and later plug-in)
Infinity Connect desktop client	• Allows a participant to join a Virtual Meeting Room or Virtual Auditorium, or make a call via the Pexip Distributed Gateway, using a lightweight client on any PC with any operating system. • Allows users to register their clients in order to receive incoming calls. Supported on: • Microsoft Windows 7 and later • Mac OS X 10.7 and later • Ubuntu Linux

Feature	Description
Infinity Connect Mobile client	• Allows a participant to join a Virtual Meeting Room or Virtual Auditorium, or make a call via the Pexip Distributed Gateway, using a client downloaded onto their mobile device. • Enables participants to view presentations on their mobile device, regardless of whether they are a video, audio-only, or control-only participant. • Users can register their device in order to receive incoming calls. Available versions: • Infinity Connect Mobile client for iOS • Infinity Connect Mobile client for Android

Audio and video specifications

Feature	Description
Supported protocols	• H.323 • SIP • WebRTC • RTMP • Microsoft Lync / Skype for Business • Individual protocols can be administratively enabled and disabled.
Audio codecs	• G.711(a/μ) • G.719 (this product is covered by patent rights licensed from Telefonaktiebolaget LM Ericsson) • G.722 • G.722.1, G.722.1 Annex C (licensed from Polycom®) • Siren7™, Siren14™ (licensed from Polycom®) • G.729, G.729A, G.729B • Opus • SILK • MPEG-4 AAC-LD (MPEG-4 video technology licensed by Fraunhofer IIS) • Speex • AAC-LC
Video codecs	• H.261 • H.263, H.263++ • H.264 (Constrained Baseline Profile, Baseline Profile and High Profile), H.264 SVC (UCIF Profiles 0, 1) • VP8 • VP9 (technology preview only) • Flash video (for Internet Explorer and Safari browser support) • RTVideo (licensed from Microsoft®).
Content sharing	• H.239 (for H.323) • BFCP (for SIP) • RDP (for Microsoft Lync / Skype for Business) • VbSS (for Skype for Business) (technology preview only) • PSOM (for presenting PowerPoint files from Microsoft Lync / Skype for Business clients) • VP8 (for WebRTC high framerate) • JPEG (for apps and web).

Feature	Description
Bandwidth	• Connections from 8 kbps per participant (G.729, audio-only), up to 6 Mbps per participant (will vary depending on the deployment environment, video resolutions, etc).
Other audio and video features	• Resolutions from QCIF to Full HD 1080p (1920 x 1080); 4:3 and 16:9 aspect ratios. • Frame rates up to 30 fps. • Customizable video watermarking. • Pexip StudioSound™ for recording-studio audio quality. • Wideband audio mixing. • Automatic gain control. • Control individual audio via Infinity Connect clients. • Support for AES (128-bit key size), DTLS SRTP, and H.235 for H.323 media encryption.

Host hardware requirements

Feature	Description
CPU	**Conferencing Nodes** • We recommend Intel Xeon E5-2600 series (Haswell architecture) or similar Xeon processors from 2012 or later, 2.3 GHz or faster. • We recommend 10-12 physical cores per socket. **Management Node** • Any processor, 2.0 GHz or faster. • 2 cores.
RAM	**Conferencing Nodes** 1 GB RAM per vCPU, so either: • 1 GB RAM per physical core (if deploying 1 vCPU per core), or • 2 GB RAM per physical core (if using hyperthreading and NUMA affinity to deploy 2 vCPUs per core). **Management Node** • Minimum 4 GB RAM for the Management Node.
Storage	**Conferencing Nodes** • 50 GB minimum per Conferencing Node • 500 GB total per server (to allow for snapshots etc.) **Management Node** • 100 GB SSD
GPU	• Host servers do not require any specific hardware cards or GPUs.
OS	• The Pexip Infinity VMs are delivered as VM images (.ova etc.) to be run directly on the hypervisor. No OS should be installed.
Network	• Gigabit Ethernet connectivity is strongly recommended. • In general, you can expect 0.5-3 Mbps per call, depending on call control setup.
Multiple VMs sharing the same hardware	• Pexip Infinity Conferencing Nodes and Management Nodes may share the same physical host. • Pexip nodes may also share the same physical host with other virtual machines. • Pexip virtual machines must be configured with dedicated CPU and memory resources, i.e. Pexip virtual machines do not support oversubscription.

Feature	Description
Service provider considerations	A Pexip deployment can manage multiple customers in various ways: • **Single Management Node, multiple domains, shared Conferencing Nodes** A single installation with one Management Node and one or more Conferencing Nodes is used by all customers. Call control or DNS sends calls for all domains to the shared Conferencing Nodes. Does not provide dedicated capacity per customer. • **Single Management Node, multiple domains, dedicated Conferencing Nodes** One or more Conferencing Nodes per customer. Allows for dedicated capacity per customer. • **Dedicated Management Node and dedicated Conferencing Nodes per customer instance** Allows for close customer network integration, using VLANs, hosted on a shared server farm with multiple VLANs. The dedicated Management Node allows for customer self-management.

Capacity

Feature	Description
Call capacity	Capacity is dependent on server specifications. As a general indication, using our recommended hardware (Intel Haswell, 10 cores, 2.3 GHz) Pexip Infinity can connect: • up to two High Definition 720p30 calls per CPU core (based on 1.1 GHz per call plus 20% headroom) • up to 20 audio-only AAC-LD calls at 64 kbps. Servers that are older, have slower processors, or have fewer CPUs, will have a lower overall capacity.

Hypervisor requirements

Feature	Description
VMware	• Version 14.1 of the Pexip Infinity platform supports VMware vSphere ESXi 5.x and 6.0, although we recommend ESXi 5.5 or 6.0. Support for ESXi 4.1 is being deprecated - if you have upgraded from a version prior to v12, you can still deploy Conferencing Nodes to servers running ESXi 4.1; however if you have a new deployment using v12 or later and attempt to deploy a Conferencing Node to a server running ESXi 4.1, that node will go straight into maintenance mode. • We recommend at least the **Standard** edition. • The **Enterprise** and **Enterprise Plus** editions have additional features that can be taken advantage of by Pexip Infinity in larger deployments. • The Pexip Infinity platform will run on the **free edition** of vSphere Hypervisor. However, this edition has a number of limitations that mean we do not recommend its use except in smaller deployments, or test or demo environments. Notably, automatic deployment of Conferencing Nodes is not supported.
Microsoft Hyper-V	• The Pexip Infinity platform supports Microsoft Hyper-V in the form of Microsoft Hyper-V Server 2012 and later, and Windows Server 2012 and later.
KVM	• Pexip Infinity requires your KVM environment to include Linux kernel 3.10.0 or later, and QEMU 1.5.0 or later. This means the following distributions: Debian 8, RHEL 7, SLES 12, or Ubuntu 14.04 (or later, where appropriate).
Xen	• Pexip Infinity requires Xen 4.2 and later.
Other hypervisors and orchestration layers	• Conferencing Nodes can be provisioned with a configuration document generated independently of a generic VM image. This permits deployment of Pexip Infinity onto unsupported hypervisors as well as onto supported hypervisors that are managed by an orchestration layer. • Pexip Infinity can be deployed on Amazon Web Services (AWS). • Pexip Infinity can be deployed on Microsoft Azure. • Pexip Infinity can be deployed on the HPE Helion Openstack® Cloud platform.

What's new in version 14.1?

The new features and enhancements and changes in functionality included in Pexip Infinity version 14.1 are described below.

For full information about this release, see the release notes.

For information about earlier versions of Pexip Infinity, see Features added in previous releases.

New features

Pexip Infinity platform

Feature	Description	More information
Layout improvements	• We have increased the relative size of video images and reduced the amount of black space. • Thumbnails of participants entering the stage will now fade into place (previously they slid in from the side).	Selecting the layout seen by participants
Show main speakers' names	You can now configure Virtual Meeting Rooms and Virtual Auditoriums to show the name(s) of the participant(s) who are being shown in the main video. Names will be displayed at the bottom of the participant's video.	Showing the names of speakers
Live platform status overview	Pexip Live View (**Status > Live View**) is a new interactive graphical report that shows the current status of your deployment, including load on Conferencing Nodes and conferences taking place. Minor changes have been made to the conference graph to align it with the new Live View.	Viewing live platform status Viewing current conference status
Test call service	Pexip Infinity now includes a new **Test Call Service**. This provides a test loop that allows users to check the quality of their video and audio, and verifies that they can connect to the Conferencing Node. On upgrade to version 14, a single default test service with the name "Test Call Service" and the alias "test_call" is automatically added to your system configuration. (If an existing conference or alias exists with those names, they will not be reassigned to the new service.) You can add further test services and aliases as necessary, and configure them via themes if required.	Configuring the Test Call Service
Certificate Signing Requests (CSRs)	You can generate a Certificate Signing Request (CSR) from within Pexip Infinity, and then upload the returned certificate associated with that request.	Certificate signing requests (CSRs)
Allow only registered devices to make Pexip Distributed Gateway calls	You can configure Call Routing Rules to only allow devices that are registered to Pexip Infinity to be able to make calls via the Pexip Distributed Gateway. This provides greater security and control over the devices that can use gateway resources.	Configuring Call Routing Rules
Local policy scripts	Local policy allows you to manipulate service configuration and media location data (that has been provided either via the external policy API, or has been retrieved from Pexip Infinity's own database) by running a jinja2 script against that data. Local policy and external policy are configured via policy profiles (**Call Control > Policy Profiles**) and a policy profile is than assigned to a system location. On upgrade to version 14, any existing external policy servers are converted into policy profiles.	**Using external and local policy to control Pexip Infinity behavior**
Dual network interfaces (NICs) for Conferencing Nodes	You can configure a secondary network address on a Conferencing Node. If configured, this secondary address is used for signaling and media, and the primary address is used for inter-node communication.	Conferencing Nodes with dual network interfaces (NICs)

Feature	Description	More information
Improved CSV import/export	The existing CSV import/export utility for Virtual Meeting Rooms has been expanded to include all configuration options. Additional CSV import/export is now available for Virtual Auditoriums, Virtual Receptions, Automatically Dialed Participants, and device aliases.	Bulk import/export of service configuration Bulk import/export of Automatically dialed participants (ADPs) Bulk import/export of device aliases
H.264 High Profile codec support	Pexip Infinity now supports the H.264 High Profile codec. The H.264 High Profile codec can provide HD video quality with reduced bandwidth requirements on H.264 connections. This codec is disabled by default. To enable the codec, go to **Platform Configuration > Global Settings** and, in the **Codecs** section, move ***H.264 High (mode 1)*** into the list of **Chosen Codecs**. For optimal interoperability results, only enable ***H.264 High (mode 1)*** — leave ***H.264 High (mode 0)*** in the **Available Codecs** list.	About global settings
RTMP dual stream support for Automatically Dialed Participants (ADPs)	Automatically Dialed Participants (ADPs) can be configured with RTMP dual streams.	Automatically dialing out to a participant from a conference
Skype for Business Video-based Screen Sharing (VbSS) *	Pexip Infinity now supports sending and receiving content to and from Skype for Business clients using VbSS. This is a "tech preview" feature and is currently only supported when the Skype for Business client is either calling another endpoint via the Pexip Distributed Gateway, or calling into a Virtual Meeting Room or Virtual Auditorium. Default: disabled.	Enable Lync / Skype for Business Video-based Screen Sharing (VbSS)
New alarm	Pexip Infinity will raise an alarm if it detects that the underlying hardware appears to be over-committed.	Viewing alarms

* Technology preview only

Infinity Connect Web App

Following are the changes to the Infinity Connect Web App in Pexip Infinity version 14:

Feature	Description	More information
Screensharing quality setting	Prior to joining a conference, you can set the frame rate to use when sharing your screen (supported in Chrome only). A lower frame rate will result in ***sharper*** images and is best for static presentations; a higher frame rate will be less sharp and is best for content where there is more ***motion***.	
Screensharing tabs	When sharing your screen (supported in Chrome only), the available desktops and apps now appear in separate tabs.	Using Infinity Connect in-call controls

Changes in functionality in this release

Feature	Description	More information
Less verbose **support.sip** messages in the support log	Some **support.sip** messages in the support log are less verbose than previously. Log messages for SUBSCRIBE, NOTIFY, OPTIONS, INFO and UPDATE requests now generate summary output.	

Feature	Description	More information
Keep alive pings from SIP proxies no longer trigger external policy requests	Pexip Infinity now immediately responds with 200 OK to OPTIONS keep alive pings from remote SIP proxies, and does not perform any additional processing — such as sending outbound external policy requests — on receipt of such messages.	
Lync and Skype for Business terminology updates	References to "Lync" on the Pexip Infinity Administrator interface have been updated to also refer to "Skype for Business", specifically: • "Lync servers" are now called "Lync / Skype for Business servers". • The "Lync (MS-SIP)" protocol (used in Call Routing Rules and when dialing out) is now called "Lync / Skype for Business (MS-SIP)". • In Call Routing Rules the "Lync AVMCU conference direct (not via Virtual Reception)" call target is now called "Lync / Skype for Business meeting direct (not via Virtual Reception)", and the "Lync clients or Lync AVMCU conferences via a Virtual Reception" call target is now called "Lync / Skype for Business clients, or meetings via a Virtual Reception". • The "Lync MSSIP domain" (used in global settings and system location configuration) is now called "Pexip Infinity domain (for Lync / Skype for Business integration)". • Global setting "Enable Lync auto-escalation" is now called "Enable Lync / Skype for Business auto-escalation". • "Lync conference lookup location" on a Virtual Reception is now called "Lync / Skype for Business meeting lookup location". (Note that all API parameter/field names are unchanged.)	
Management API improvements	There have been minor updates to the management API: • It now allows filtering of devices ('device' resource) by the owner email address ('primary_owner_email_address' field).	Retrieving, paginating, filtering and ordering resource details
Improvements to the conference status graph	There have been minor updates to the conference status graph, including changes to the icons representing Conferencing Nodes, and the inclusion of control-only participants in the graph.	Viewing current conference status
More historical media load data	The Conferencing Node media load graph now shows 4 weeks of history (rather than one week as previously).	Viewing Conferencing Nodes
Relationship between services and themes	You can now: • from each of the Service configuration pages, view which themes are associated with the services, and sort the list by theme • view the number of services to which a particular theme has been applied.	Viewing the themes currently in use
Service creation time	You can now view the time at which a particular service was created. This information is shown via the Administrator interface on the edit page for each service. This applies to: • Virtual Meeting Rooms • Virtual Auditoriums • Virtual Reception • Automatically Dialed Participants • Device aliases	
Maintenance mode status	The Conferencing Node status page now includes a sortable column regarding whether each node is in maintenance mode.	
Conference history	The maximum number of conferences show in the **Conference History** page has increased from 5,000 to 10,000.	Viewing historical information about conferences

Feature	Description	More information

Security-related updates	This release includes several security-related modifications:
	• TLS connections no longer use Triple DES or MD5.
	• The security wizard (used when deploying Pexip Infinity in secure mode):
	  ○ allows you to disable TLS 1.0
	  ○ allows you to disable Anonymous Diffie Hellmann key exchange for outbound SIP/TLS
	  ○ no longer allows you to enable SSL 3.0.
		Deploying Pexip Infinity in a secure mode of operation

Infinity Connect Web App support for Microsoft Internet Explorer on Windows 10 has been deprecated.	Support for the Infinity Connect Web App via Microsoft Internet Explorer on Windows 10 systems has been deprecated. Windows 10 users should use Microsoft Edge (or another browser such as Chrome, Firefox or Opera) instead.	
API participants and backplane reservation	If an API participant is the first participant to join a conference, it will now reserve a backplane for the conference. Previously an API participant would not have reserved a backplane.	

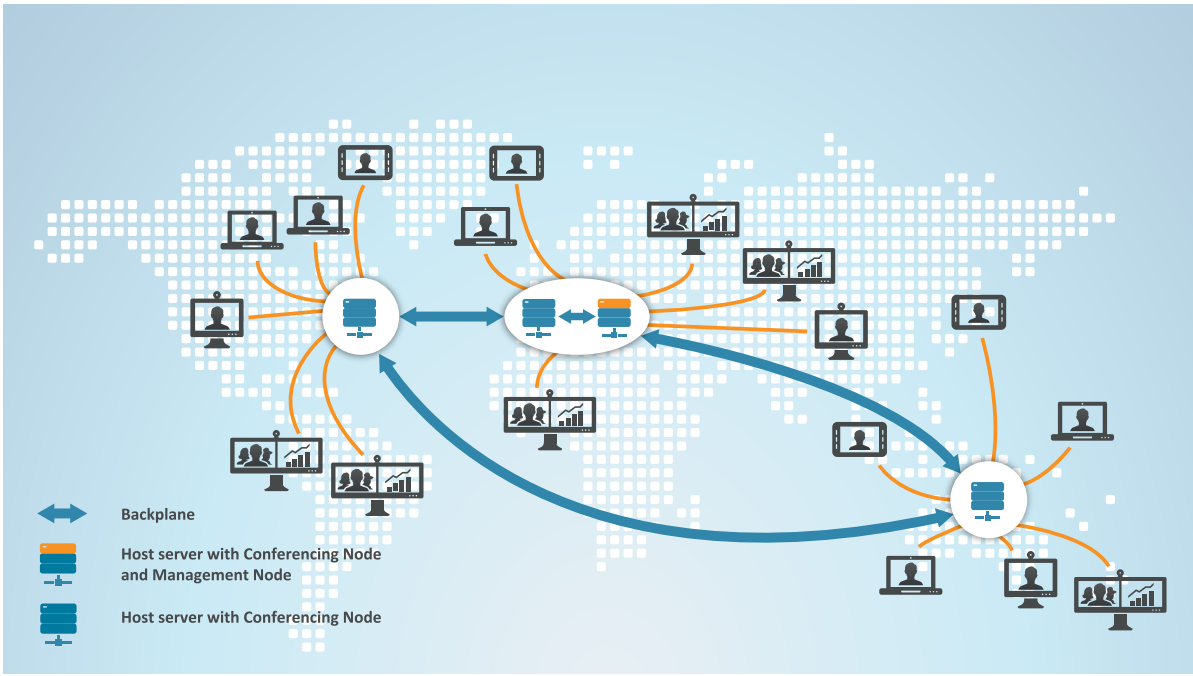
Pexip Infinity components

The Pexip Infinity conferencing platform is a virtual entity that consists of a **Management Node** and one or more securely interconnected **Conferencing Nodes**. Both are software applications that you deploy as Virtual Machines (VMs) on **host servers** distributed around the globe. You can add, remove or move Conferencing Nodes according to your conferencing requirements.

Conferences take place in **Virtual Meeting Rooms** and **Virtual Auditoriums**, which each having one or more associated **aliases**. Conference participants access a Virtual Meeting Room or Virtual Auditorium by dialing any one of its aliases directly, or via the **Virtual Reception** IVR service. This connects them to the Virtual Meeting Room or Virtual Auditorium on their nearest Conferencing Node. A single such conference can take place across one, two, or more Conferencing Nodes with no difference in conference experience from the participants' perspective.

Conference participants can access Virtual Meeting Rooms and Virtual Auditoriums from virtually any endpoint, including the **Pexip Infinity Connect** suite of clients (which includes the Infinity Connect desktop client, the Infinity Connect Mobile client and the Infinity Connect Web App). Infinity Connect clients can also be used to control the conference, view presentations, share content, and chat with other conference participants.

The **Pexip Distributed Gateway** service interworks calls between protocols, allowing users to make calls from virtually any type of endpoint (including SIP, H.323, Lync / Skype for Business) and Infinity Connect clients (WebRTC and RTMP).



Pexip Infinity deployment showing Management Node and four Conferencing Nodes with participants connected locally

Management Node

The Management Node is the administrative interface of the Pexip Infinity platform. It hosts the Pexip Infinity Administrator interface, from which administrators can:

- Create and manage Conferencing Nodes.
- Create and configure Pexip Infinity services (Virtual Meeting Rooms, Virtual Receptions and so on).
- View platform and conference status across all Conferencing Nodes.
- Perform active conference management functions such as dialing out to new participants, adding streaming participants, locking a conference, muting a participant's audio and disconnecting participants. However, the Management Node does not handle any conference media or signaling.

It is deployed using a virtual machine management application such as VMware's vCenter Server, or Microsoft Hyper-V, or on a cloud service such as Amazon Web Services or Microsoft Azure.

Conferencing Nodes

The Conferencing Nodes provide the capacity for conferences.

- They handle all conference media and signaling.
- There is no limit on the number of Conferencing Nodes that you can add to the Pexip Infinity platform.
- All Conferencing Nodes get the same service configuration from the Management Node. This means that participants throughout your organization can access the same Pexip Infinity services (Virtual Meeting Rooms, Virtual Receptions and so on) even though they might be connected to different Conferencing Nodes.
- Conferencing Nodes are deployed via the Management Node. A Conferencing Node can be deployed fully automatically on host servers running VMware. On host servers running on other hypervisors or when running in a cloud service, you use the Management Node to configure the new Conferencing Node and then complete the deployment manually using the appropriate hypervisor or cloud-provider tools.
- The Pexip Infinity platform can have Conferencing Nodes that are deployed on one or more host servers, across one or more system locations and managed by one or more types of hypervisor, or it can be a hybrid deployment with nodes running on a combination of on-premises and cloud-hosted servers. A Conferencing Node can co-exist on the same host server as a Management Node.
- Conferencing Nodes can be deployed with dual network interfaces.

Pexip Infinity Connect clients

Conference participants do not need to have a traditional video endpoint in order to access Pexip Infinity services.

The complementary Pexip Infinity Connect suite of clients allow users to connect to any Virtual Meeting Room or Virtual Auditorium within the Pexip Infinity deployment, either:

- directly from a web browser without any special downloads or plugins
- from an installable desktop client
- from an Infinity Connect Mobile client, available for iOS or Android.

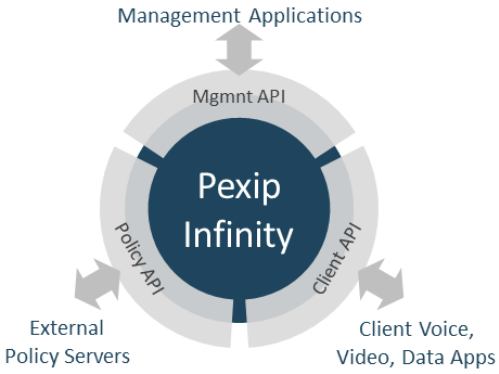
In addition to connecting with video and audio, Infinity Connect users can control the conference, view presentations, share content and chat. Infinity Connect can also be used to make direct calls to other devices or systems when used in conjunction with the Pexip Distributed Gateway.

For more information on using and administering Infinity Connect, see Introduction to Infinity Connect.

Pexip Infinity APIs and SDKs

Pexip Infinity incorporates several powerful and comprehensive APIs:

- **Management API**: a REST API used for configuring the entire Pexip Infinity deployment, viewing history and status, and issuing commands. See Introduction to the management API for more information.
- **Client API**: a REST API used for managing calls and participants, such as connect, disconnect, mute and unmute, presentation controls, DTMF, etc. See Pexip client REST API for more information.
- **Policy API**: a REST API used to defer decision-making to external policy servers instead of using the built-in call policies within Pexip Infinity. See **Using external and local policy to control Pexip Infinity behavior** for more information.



In addition to these REST APIs, a Javascript API is also available for building custom web-based clients. See PexRTC JavaScript client API for more information.

Virtual Meeting Rooms and Virtual Auditoriums

Conferences take place in Virtual Meeting Rooms (VMRs) and Virtual Auditoriums (a type of VMR that is optimized for use by a small number of hosts and a large number of guests). Virtual Meeting Rooms and Virtual Auditoriums can host any number of people from

any type of device, subject to any participant limits that have been applied by administrator.

When creating Virtual Meeting Rooms and Virtual Auditoriums, you define the aliases that are used to access the Virtual Meeting Room, and any PINs that are required.

All Virtual Meeting Rooms and Virtual Auditoriums exist on all Conferencing Nodes. When a Conferencing Node receives a call to a particular Virtual Meeting Room or Virtual Auditorium alias, it creates a conference instance based on that service's settings. In this way, resources are not used until the first caller actually places a call into the conference.

When a second endpoint places a call to an alias that belongs to the same Virtual Meeting Room or Virtual Auditorium, the call is placed into the existing conference instance.

There is no limit on the number of VMRs that can be configured on your Pexip Infinity platform. VMRs do not consume resources on your deployment unless they are actually being used to host a conference. Unless you manually restrict access, the number of participants who can access a particular VMR, and the number of VMRs that can be in use at the same time, are limited only by the size of your Pexip Infinity deployment (in terms of server capacity and licenses).

VMRs can be bulk-provisioned from directory information contained in a Windows Active Directory LDAP server, or any other LDAP-accessible database.

For more information, see About Virtual Meeting Rooms and About Virtual Auditoriums.

Virtual Receptions

A Virtual Reception allows you to set up one or more direct dial telephone numbers or IP addresses that participants can dial to access a single, central IVR service. From here, participants can use DTMF tones to select the Virtual Meeting Room or Virtual Auditorium they wish to join. It provides an alternative means to access Pexip Infinity services for participants who cannot dial aliases directly.

For more information, see About the Virtual Reception IVR service.

Pexip Distributed Gateway

The Pexip Distributed Gateway service interworks calls between protocols, allowing users to make calls from virtually any type of endpoint (including SIP, H.323, Lync / Skype for Business) and Infinity Connect clients (WebRTC and RTMP) and interworking these to SIP, H.323 or Lync / Skype for Business.

For more information, see About the Pexip Distributed Gateway service.

Test Call Services

Pexip Infinity provides a test loopback service that allows users to check the quality of their video and audio (i.e. that their local camera, microphone and speakers are working properly), and verifies that they can connect to a Conferencing Node.

For more information, see Configuring the Test Call Service.

Host servers

The Management Node and Conferencing Nodes are virtual machines (VMs) that run on industry-standard host servers. A Management Node can run on the same host server as a Conferencing Node. Other Conferencing Nodes can run on host servers in the same or different locations, allowing you to create a globally distributed system.

You can have two Conferencing Nodes running on the same host server, for example to ensure service continuity during upgrade of one of the Conferencing Nodes. However, you must ensure that your hardware is not overcommitted - see Detailed server hardware requirements for more information.

The Pexip Infinity platform can also be deployed as a cloud service via **Amazon Web Services (AWS)** or **Microsoft Azure**, with private, public or hybrid deployment options.

Hypervisors

Each host server runs a hypervisor, an application which manages virtual machines and the physical hardware on which they are hosted. Pexip Infinity version 14.1 includes specific support for the following hypervisors:

- VMware vSphere ESXi (4.1*, 5.x and 6.0)
  * Support for ESXi 4.1 is being deprecated
- Microsoft Hyper-V (Microsoft Hyper-V Server 2012 and later, or Windows Server 2012 and later)
- KVM
- Xen (4.2 and later)

Other hypervisors and orchestration layers may be used but are not officially supported. If you wish to deploy Pexip Infinity using a non-supported hypervisor, we recommend that you contact your Pexip authorized support representative for assistance.

VM Managers

A VM manager is an application that allows you to connect to one or more VMware vSphere ESXi hypervisors when Automatically deploying a new Conferencing Node on a VMware host. VM managers supported by version 14.1 of Pexip Infinity are vCenter Server and vSphere.

- **vCenter Server** is an application used to manage groups of host servers (and therefore ESXi hypervisors) through a single interface.
- **vSphere** on the host server is used when managing a single host server (and therefore ESXi hypervisor).

For more information, see Using a VM manager.

Endpoints

You can register SIP and H.323 endpoints to Pexip Infinity, alternatively your endpoints could register with a call management system.

For more information, see Registering devices to Pexip Infinity and DNS record examples.

Call control

Supported call control solutions

Pexip Infinity can be easily integrated with virtually any existing SIP, H.323 and Lync / Skype for Business call control solutions including Cisco UCM, Cisco VCS, Polycom CMA, Polycom DMA, Avaya Aura, Microsoft Lync 2010 and 2013, Skype for Business, and others.

Local and external policy

You can extend Pexip Infinity's built-in functionality by using external and/or local policy to apply bespoke call policy and routing decisions based on your own specific requirements.

See **Using external and local policy to control Pexip Infinity behavior** for more information.

Gatekeeper registration

Pexip Infinity does not register with a gatekeeper as an MCU.

Using a gatekeeper to route calls to Pexip Infinity conferences

To ensure that calls can be routed to Pexip Infinity, your gatekeeper or call control system must be configured with appropriate neighbor/zone relationships towards the Pexip Infinity Conferencing Nodes. These zones must be set up so that when an endpoint places a call to a Pexip Infinity alias, the call is routed to the endpoint's local Conferencing Node(s) as a first preference. Other non-local Conferencing Nodes can be used as secondary choices to provide redundancy.

For further information about how to configure your specific call management system to work with Pexip Infinity, see the following documentation:

- Pexip Infinity and Microsoft Lync / Skype for Business Deployment Guide
- Pexip Infinity and Cisco VCS Deployment Guide
- Pexip Infinity and Cisco Unified Communications Manager Deployment Guide
- Pexip Infinity and Polycom DMA Deployment Guide

Distributed architecture

Benefits and components of the Pexip Infinity distributed architecture

Pexip Infinity is built on top of a distributed architecture which provides:

- centralized management of any number of Conferencing Nodes in any number of locations
- ability to deploy conferencing resources where and when required, without service outage
- significant WAN bandwidth savings in conferences that span locations
- consistent user experience, independent of the number of Conferencing Nodes
- increased resilience to temporary network outages
- ability to use Conferencing Nodes as gateways for point-to-point calls, thus avoiding hairpinning of media back to a centralized datacenter
- allocation of licenses from a central pool.

Distributed architecture components

Pexip Infinity distributed architecture is purely software-based and virtualized, running on industry-standard servers. It consists of:

- a single Management Node. A Pexip Infinity deployment, regardless of size, has just one Management Node. The purpose of the Management Node is to create and manage Conferencing Nodes. The Management Node is in neither the signaling nor the media path of a conference.
- one or more Conferencing Nodes. A minimal Pexip Infinity deployment will have one Management Node and one Conferencing Node. However, most Pexip Infinity deployments will have multiple Conferencing Nodes.

Configuration and provisioning data is pushed out from the Management Node to all the Conferencing Nodes; diagnostics data and event information is sent from the Conferencing Nodes back to the Management Node. There is never a need for the administrator to manage a Conferencing Node directly. This centralized management ensures that the entire deployment is configured with a consistent data set – the entire deployment acts as a single application.

A Management Node can run on the same host server as a Conferencing Node. Other Conferencing Nodes can run on host servers in the same or different locations, allowing you to create a globally distributed system. You can have two Conferencing Nodes running on the same host server, for example to ensure service continuity during upgrade of one of the Conferencing Nodes, and for maximum performance.

Scaling up

An administrator can easily scale a deployment up by creating several Conferencing Nodes in the same location (i.e. the same datacenter). Capacity can even be added “on the fly” – Conferencing Nodes can be added in a couple of minutes if more capacity is needed.

Scaling out

A typical Pexip Infinity deployment consists of two or more locations. If a customer has three main offices such as New York, London and Tokyo, and a concentration of users in those locations, it would make sense to deploy Conferencing Nodes in all three locations. There is no limit on the number of locations in a Pexip Infinity deployment. When additional locations need to be added, this can be done “on the fly” while the system is operating, with no impact on service availability.

Application level resiliency

Application level resiliency greatly improves on a conference experience during for instance temporary network outages, as Pexip Infinity will automatically re-establish the conference when the network connection is re-established.

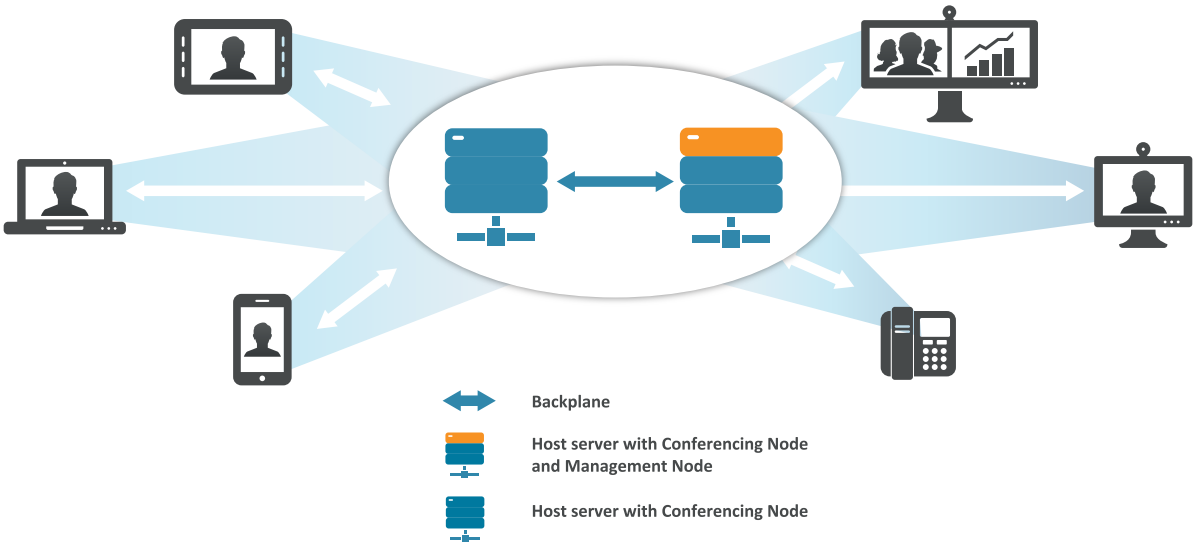
Conference distribution

All Pexip Infinity services (Virtual Meeting Rooms, Virtual Auditoriums, Virtual Receptions and the Pexip Distributed Gateway) exist on all Conferencing Nodes. When a user dials in to a Virtual Meeting Room, or Virtual Auditorium a **conference instance** is created.

This conference can exist across one or more Conferencing Nodes. Where two or more Conferencing Nodes are involved, they send the call media between each other over a secure IPsec **backplane**.

When you first deploy a Conferencing Node, you select its system location, which is a user-defined parameter that tells Pexip Infinity where the Conferencing Node is physically located. Conferences that involve more than one Conferencing Node can be locally distributed, globally distributed, or both, depending on the system location of each of the Conferencing Nodes involved. Each of these distribution scenarios are described in more detail below.

Locally distributed conferences



Locally distributed conference with two Conferencing Nodes across two host servers in the same location

Locally distributed conferences exist across two or more Conferencing Nodes in the same physical location (i.e. nodes with the same **System location** selected) as shown in the diagram above.

Having more than one Conferencing Node in a single location allows you to increase capacity and provide redundancy.

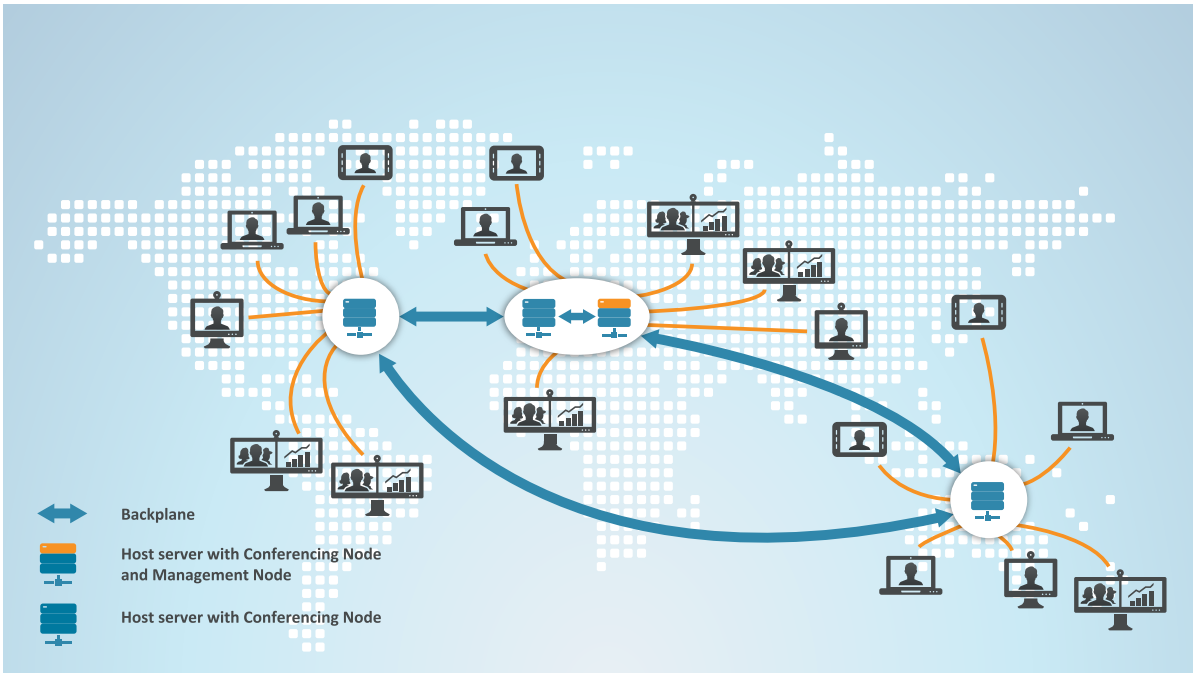
To maintain efficiency, no more than three Conferencing Nodes in a single location will handle the media for a particular conference instance. However, you can configure "overflow" locations that will handle the media when the original location reaches its capacity for a conference instance. For more information, see Handling of media and signaling and Location capacity overflow.

Globally distributed conferences

Globally distributed conferences exist across two or more Conferencing Nodes in different physical locations (i.e. nodes with a different **System location** selected). This allows participants in different regions to access the conference from their local Conferencing Node. Call media is sent between the Conferencing Nodes on behalf of all the endpoints connected to them, thus minimizing WAN bandwidth usage between locations.

This helps to provide a superior meeting experience as it reduces the distance that the media has to travel between the endpoint / client and the Conferencing Nodes.

Locally and globally distributed conferences



Locally and globally distributed conference with four Conferencing Nodes across four host servers in three locations

Conferences can also be **locally and globally distributed** at the same time, if two or more Conferencing Nodes in one location and at least one other Conferencing Node in a different location are involved. In such cases, one Conferencing Node in each location will act as the **proxy** for any other Conferencing Nodes in the same location that are handling the media for that conference. Call media for each location is sent between the proxies only, thus minimizing WAN bandwidth usage between locations.

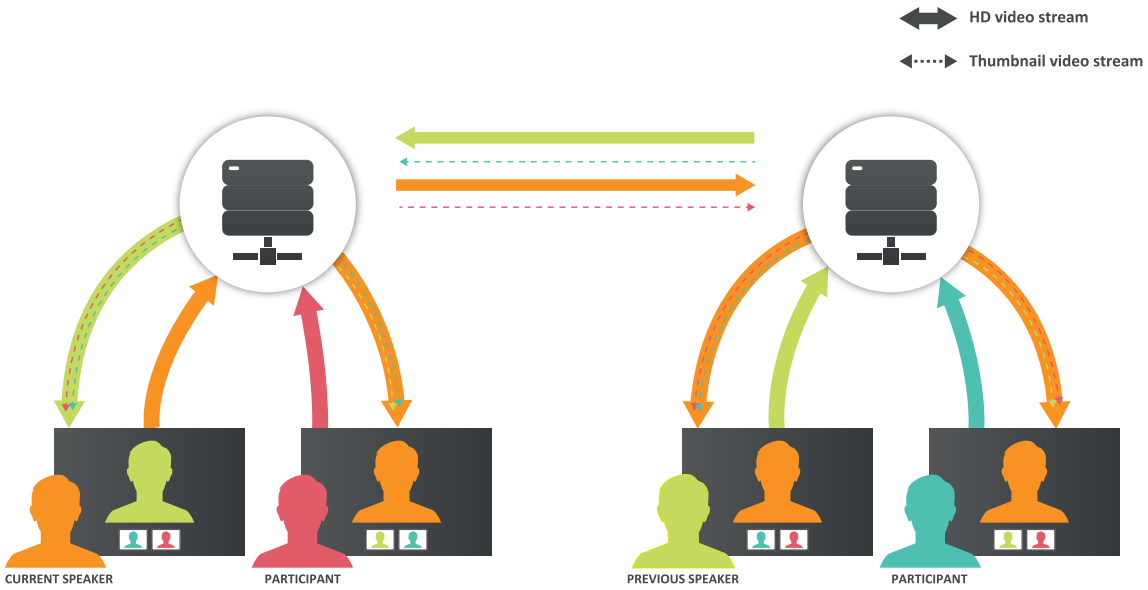
Bandwidth management

Bandwidth usage

The Conferencing Node to which the current speaker is connected sends an HD video stream of the current speaker, plus a lower-resolution thumbnail (a smaller image at the bottom of the screen which shows the participant's video) of each of the other participants connected to it.

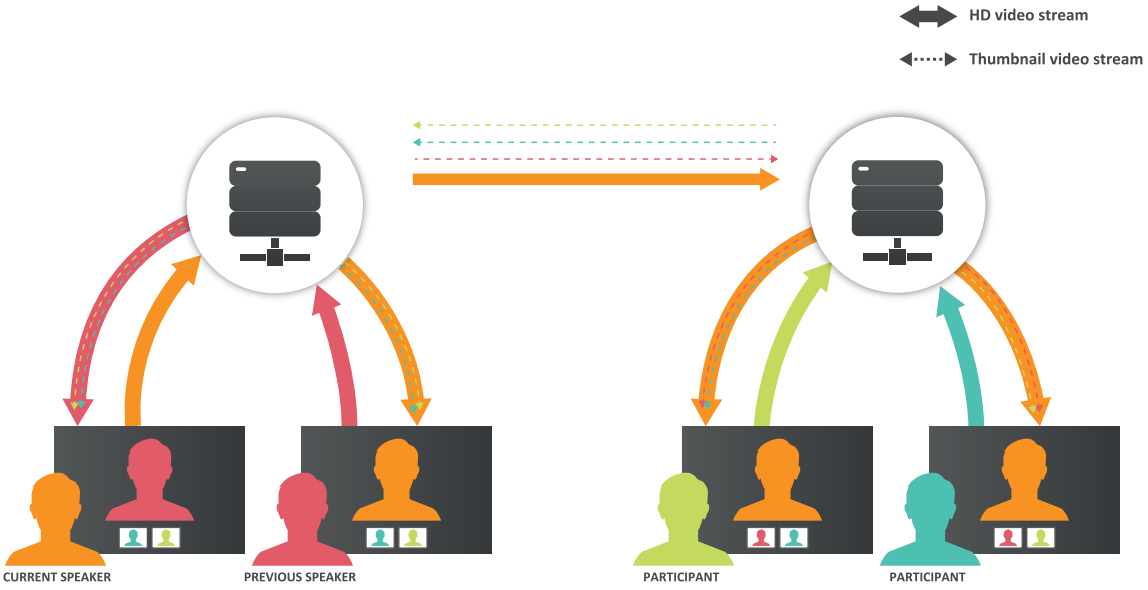
The Conferencing Node to which the previous speaker is connected sends an HD video stream of the previous speaker, plus lower-resolution thumbnails of each of the other participants connected to it.

All other Conferencing Nodes send lower-resolution thumbnails of each of the participants connected to it.



Bandwidth usage when current and previous speakers are connected to different Conferencing Nodes

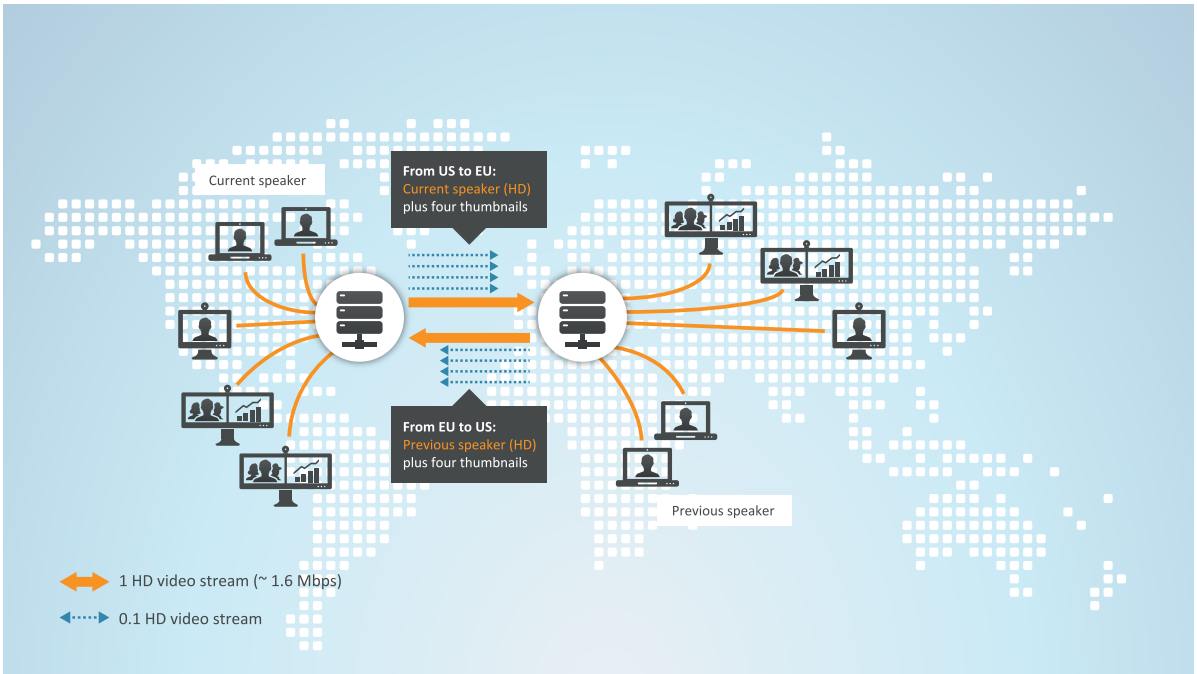
As shown in the diagram above, if the current and previous speakers are connected to different Conferencing Nodes, there is one HD stream in each direction between the two nodes, plus one smaller stream for every participant being shown in a thumbnail.



Bandwidth usage when current and previous speakers are connected to the same Conferencing Node

As shown in the diagram above, if the current and previous speakers are connected to the same Conferencing Node, that Conferencing Node sends an HD video stream of the current speaker plus lower-resolution thumbnails of each of the other participants, to all other Conferencing Nodes. The other Conferencing Nodes send lower-resolution thumbnails of the other participants.

This architecture means that there are major bandwidth savings when compared with traditional MCU deployments where all conference participants regardless of location connect to the same MCU and individual HD and thumbnail video streams are sent between the MCU and every endpoint.



Significant bandwidth savings are achieved due to fewer and smaller video streams being sent between locations.

Note that:

- One HD stream uses about 1.6 Mbps.
- One thumbnail stream uses about 64-192 kbps.
- A presentation stream (not shown in the diagrams above) uses an additional 1.6 Mbps from the presenter.
- Pexip Infinity supports up to 6 Mbps per participant (this will vary depending on the deployment environment, video resolutions, etc).

Downspeeding

Pexip Infinity will automatically downspeed and upspeed individual calls in response to fluctuating network conditions.

Bandwidth restrictions

Administrators can individually restrict the bandwidth available to participants accessing each Virtual Reception, Virtual Meeting Room and Virtual Auditorium, and per Call Routing Rule. Restrictions can also be applied across an entire deployment. For more information, see Restricting call bandwidth.

Load balancing, redundancy and scalability

This topic describes the load balancing, redundancy and scalability aspects of Pexip Infinity.

Load balancing

Pexip Infinity load-balances intelligently across all Conferencing Nodes that are grouped within a location (**Platform Configuration > Locations**), according to the following logic:

- Signaling is always handled by the node that originally receives the call. Signaling cannot be moved, and always remains routed to the original node in the local location.
- If a node in that location is already handling media for this conference, and it has spare capacity, then that node will also handle the media on the new call.

- If no nodes are handling media for this conference, or all nodes currently handling media for this conference have no spare capacity, then media processing will occur on the least loaded node in the location. (Note that no more than three nodes per location can handle the media for a particular conference instance.)
- If more than one node is now involved in media processing, a backplane is created between the nodes.
- If a location has no spare capacity for this conference, the least loaded node in the overflow location is used, if configured.

For more information see Handling of media and signaling.

Load balancing of signaling between Conferencing Nodes in a single location is achieved via your call management system. You should configure it so that calls from each configured set of endpoints are routed to a configured set of Conferencing Nodes. You should configure DNS SRV records to identify all Conferencing Nodes in a location.

For further information about how to configure your specific call management system to work with Pexip Infinity, see the following documentation:

- **Pexip Infinity and Microsoft Lync / Skype for Business Deployment Guide**
- **Pexip Infinity and Cisco VCS Deployment Guide**
- **Pexip Infinity and Cisco Unified Communications Manager Deployment Guide**
- **Pexip Infinity and Polycom DMA Deployment Guide**

Redundancy

Management Node

The Management Node is used to create Conferencing Nodes, configure conference settings, manage licenses and collate system logs. It does not handle any call processing, so if it were to become unavailable the Pexip Infinity service would be unaffected. However:

- you would not be able to make changes to services (Virtual Meeting Rooms, Virtual Receptions and so on)
- logging would be affected: the Management Node would not receive and collate logs from the Conferencing Nodes. This can be mitigated by Using a syslog server to collate logs
- after 14 days, licenses would cease to be allocated and the Pexip Infinity service would no longer allow calls.

Conferencing Node

You should ensure that your call control system is set up so that, in the event of a Conferencing Node failing, new calls to the conference are routed to another available Conferencing Node.

If multiple Conferencing Nodes are configured in a single location, new calls to a conference can be routed by the call control to only those nodes that are currently available. To achieve this, you must configure the call control system with all possible Conferencing Nodes (either explicitly or through DNS records), and for liveness checks to be carried out.

Note that there is no live fail-over; calls in progress on a Conferencing Node that fails will be lost, and participants will need to redial the Virtual Meeting Room or Virtual Auditorium alias in order to reconnect to the conference.

Scalability

To increase the capacity of the Pexip Infinity platform, simply deploy one or more new Conferencing Nodes from the Management Node.

You may also need to increase the number of licenses.

Handling of media and signaling

Media and signaling for each call to a Pexip Infinity service (VMR, gateway call etc.) may take different paths, depending on the location of the caller, the available capacity of Conferencing Nodes, and whether any overflow locations have been configured.

A single conference instance can span any number of locations and any number of Conferencing Nodes, but with a limit of 3 nodes per location that are handling media for that conference.

In all cases, when a call is placed to a Pexip Infinity service, the call will be received by whichever Conferencing Node was selected by your call control system or routed via DNS. This Conferencing Node will always continue to handle the signaling for that call. The following sections describe how the media is routed in different situations:

- Locally distributed conferences
- Globally distributed conferences
- Location capacity overflow
- Pexip Distributed Gateway calls

Locally distributed conferences

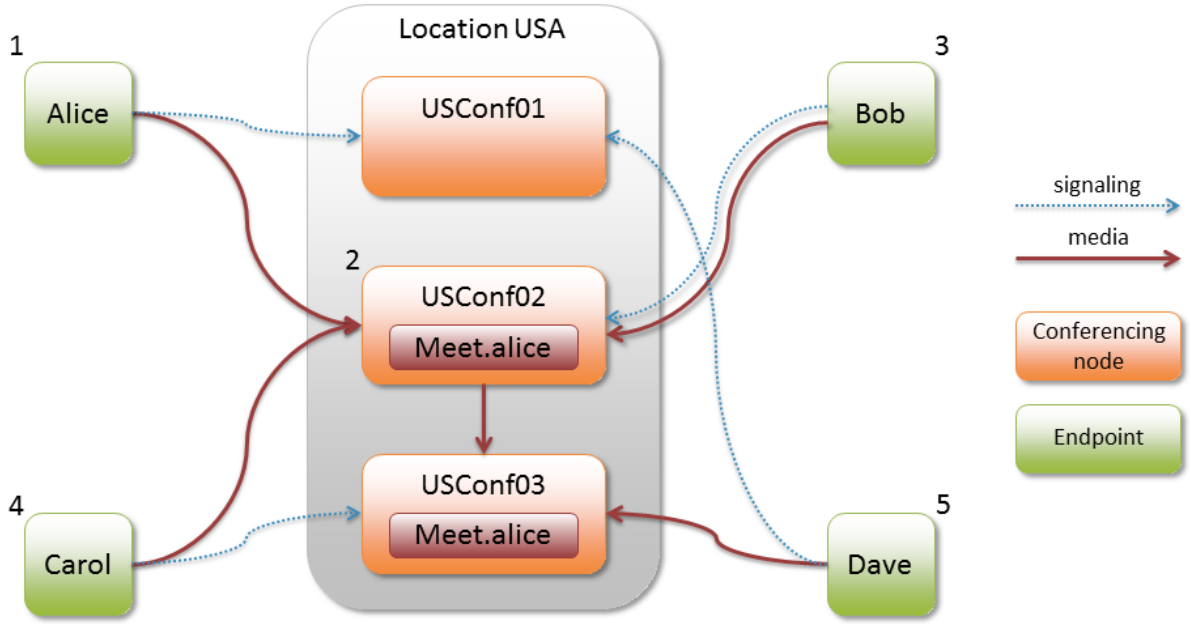
A conference is locally distributed when all participants are connected to Conferencing Nodes in a single location.

Media and signaling for a single call may take different paths within a single location, as follows:

- When an alias that belongs to a Pexip Infinity service is dialed from an endpoint, the call will be placed to whichever Conferencing Node your call control system directs the call to. This Conferencing Node will always continue to handle the signaling for that call, regardless of where the media is routed.
- If the **conference does not already exist in this Conferencing Node's location**, Pexip Infinity will check the loading of all Conferencing Nodes in the location and will allocate the conference to the Conferencing Node that currently has the most available capacity. The media for the call will go directly to that Conferencing Node, but the signaling will continue to be handled on the Conferencing Node to which the endpoint connected.
- If this is a call to a **conference already running on another Conferencing Node within the same location**, the media will be set up on the node where the conference is running, but the signaling will be handled on the node to which the endpoint connected.
- If the Conferencing Node on which the conference is running reaches capacity, Pexip Infinity will allocate a secondary (and a then a tertiary) node within that location to handle media for that conference, and automatically create a local backplane between these Conferencing Nodes so all participants are part of the same conference, with the same layout.
- Up to three Conferencing Nodes per location can be involved in media handling for one conference instance.
- Within one location, when all of the local Conferencing Nodes that can handle the media for a conference instance have reached their capacity, "overflow" locations may be utilized, thus escalating it to a globally distributed conference.

  In such cases, one Conferencing Node in each location will act as the **proxy** for any other Conferencing Nodes in the same location that are handling the media for that conference. Call media for each location is sent between the proxies only, thus minimizing WAN bandwidth usage between locations.
- When all of the local Conferencing Nodes and all of the nodes in the overflow locations that can handle the media for a conference instance have reached their capacity, no further calls can be placed to that conference from endpoints connected to nodes within that local location. Other endpoints connected to Conferencing Nodes in other locations may still connect to the conference, assuming that there is sufficient capacity in those locations.

Example



Example of media and signaling in a locally distributed conference

1. Alice dials **meet.alice@example.com** and is connected to Conferencing Node **USConf01**.
2. Pexip Infinity determines that another Conferencing Node within that location, **USConf02**, has the most capacity and so sets up the conference on that node. Alice's signaling is handled by **USConf01** and the media is being sent directly to **USConf02**.
3. Bob dials **meet.alice@example.com**. He is connected to **USConf02**. The conference is already running on that Conferencing Node, so **USConf02** handles both the media and the signaling for Bob.
4. Carol dials **meet.alice@example.com**. She is connected to **USConf03**. Her signaling is handled by **USConf03** but the media is routed directly to **USConf02**, where the conference is running.
5. Dave dials **meet.alice@example.com**. He is connected to **USConf01**. Pexip Infinity determines that **USConf02** is now out of capacity, and **USConf03** has the most available capacity. Dave's media is routed to **USConf03**, but **USConf01** continues to handle his signaling.

Globally distributed conferences

A conference is globally distributed when Conferencing Nodes in two or more different locations are involved in a single conference instance. For example, a globally distributed conference will occur when a participant in the USA location and a participant in the UK location access the same conference instance via their local Conferencing Nodes.

In such cases, Pexip Infinity will establish a "geo backplane" between the Conferencing Nodes that are handling the media for the call in each location and use this backplane to send media between the two locations.

If more than one Conferencing Node in a single location is involved in handling the media for the call (in other words the conference is also locally distributed), Pexip Infinity will nominate one of the Conferencing Nodes in that location as the proxy. The proxy will send media between locations, on behalf of itself and any other Conferencing Node in its own location that are handling the media for the call. This minimizes WAN bandwidth usage between locations.

Location capacity overflow

To maintain efficiency, no more than three Conferencing Nodes per location can handle the media for a particular conference instance. However, if all three Conferencing Nodes allocated to handle the media for that conference have reached capacity, additional "overflow locations" can be utilized. These overflow locations will handle the media for new conference participants that are connected (for signaling purposes) to nodes within the original location.

Pexip Infinity deployments also have the ability to overflow into temporary Conferencing Node resources in the Amazon Web Services (AWS) cloud (also referred to as "bursting"). The AWS cloud Conferencing Nodes instances are only started up when required and are automatically stopped again when capacity demand normalizes.

Each location can be optionally configured with a **Primary overflow location** and a **Secondary overflow location**.

- When a location reaches its capacity for a conference instance, new participants joining that conference at that location will have their media routed directly to a Conferencing Node in the **Primary overflow location**. Their signaling will remain routed to the original node in the local location.
- If the conference instance running at the **Primary overflow location** is at that location's capacity, the new participants joining the conference at the original location will have their media routed directly to a Conferencing Node in the **Secondary overflow location** instead.
- If you are using dynamic cloud bursting, your bursting location containing your overflow nodes must be a **Primary overflow location**; the **Secondary overflow location** can only be used for standard overflow i.e. to other "always on" nodes.
- If the conference instance is at full capacity at the original location, and its primary and secondary locations, then no further calls can be placed to that conference from endpoints connected to nodes within that original location.

  Note that endpoints connected to nodes at other locations may still be able to join the conference. This includes endpoints connected to the primary and secondary overflow locations that are associated with a location that can no longer accept calls for that conference (this is because each location has its own set of overflow locations that it can use for its local endpoints).
- If participants leave a conference and capacity becomes available again at some locations, existing participants' media connections are not rebalanced to their local location. However, any new participants will be able to take advantage of that freed-up capacity and join the conference at those locations.
- Location overflow is triggered whenever a location reaches its capacity for a conference instance — in other words, whenever all the Conferencing Node in that location that can take the media for that conference instance have reached capacity. You do not have to deploy three Conferencing Nodes at a location before it can overflow to a different location. Examples of when overflow could happen are:
  - if there is only one node at that location and it has reached capacity
  - if there are four nodes at that location, but the three that have been allocated the media are at capacity.
- Whenever an overflow location is utilized, a "Capacity reached in the location. Allocating the media node in the overflow location." entry is written to the administrator log. The log message indicates the conference name, the original location and the overflow location.
- When each new location is brought into the conference, Pexip Infinity nominates a single Conferencing Node in that location as the proxy node for that conference instance, and a geo backplane is established between each of the proxy nodes.
- You must ensure that endpoints in their local locations are able to route their media to the Conferencing Nodes in the overflow locations.
- Note that in very rare scenarios, depending on existing media allocation and timing, a conference could span four media nodes in the same location. For example, assume that two new participants connect at the same time to an existing conference spanning two media nodes, and that those two existing media nodes are at capacity. If the two new participants' signaling is being handled by different nodes, it is possible that each signaling node could expand the conference onto a different media node in that location, resulting in four nodes handling the conference media.

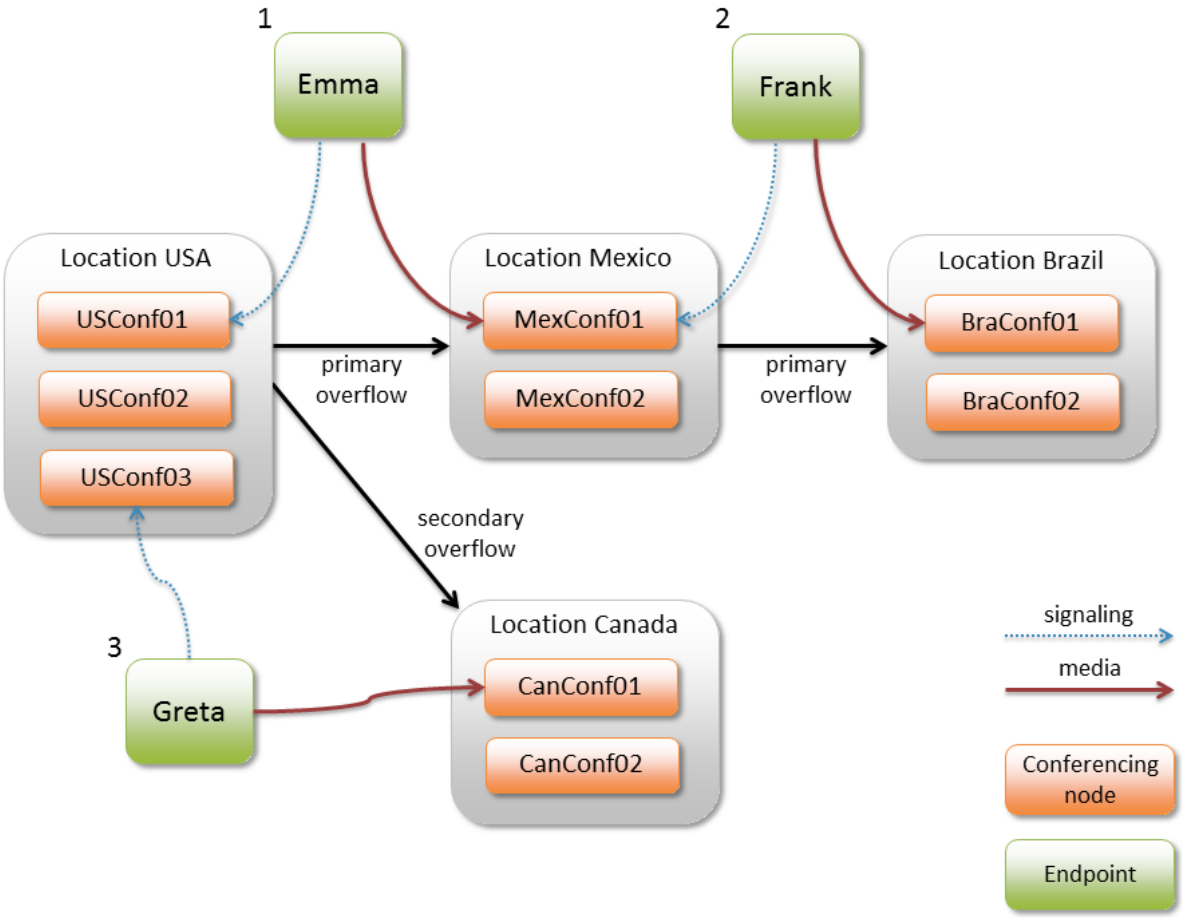
Example

This example builds on the previous example described above in Locally distributed conferences. It assumes that the conference **meet.alice@example.com** is in progress and currently has all of its media and signaling in a locally distributed conference in location **USA**.

In addition, the Pexip Infinity deployment has locations configured for USA, Mexico, Canada and Brazil, and that the USA and Mexico locations are configured with primary and secondary overflow locations as follows:

Location	Primary overflow location	Secondary overflow location
USA	Mexico	Canada
Mexico	Brazil	-

This example now shows what could happen when three new participants, two from the USA and one from Mexico, join the conference.



Example of escalating a locally distributed conference to a globally distributed conference via location overflow

1. Emma, who lives in USA, dials **meet.alice@example.com** and is connected to **USConf01** in location **USA**.
   Assume that all three Conferencing Nodes currently handling the media for this conference in location **USA** have reached their capacity, so Emma's media is routed directly to **MexConf01** in location **Mexico** (because **Mexico** is the primary overflow location for the **USA** location). **USConf01** continues to handle her signaling.
2. Frank, who lives in Mexico, dials **meet.alice@example.com** and is connected to **MexConf01** in location **Mexico**.
   Assume that all of the Conferencing Nodes currently handling the media for this conference in location **Mexico** have reached their capacity, so Frank's media is routed directly to **BraConf01** in location **Brazil** (because **Brazil** is the primary overflow location for the **Mexico** location to which Frank is connected). **MexConf01** continues to handle his signaling.
3. Greta, who lives in USA, dials **meet.alice@example.com** and is connected to **USConf03** in location **USA**.
   Assume that all of the Conferencing Nodes currently handling the media for this conference in location **USA** have reached their capacity, as have all of the nodes handling the media for this conference in the USA location's primary overflow location of **Mexico**, so Greta's media is routed directly to **CanConf01** in location **Canada** (because **Canada** is the secondary overflow location for the **USA** location to which Greta is connected; it is irrelevant if location **Brazil** has spare capacity as **Brazil** is not an overflow location for endpoints connected to the **USA** location). **USConf03** continues to handle her signaling.

Note that this example deliberately describes an extreme scenario as its purpose is to demonstrate the location overflow logic applied by Pexip Infinity.

Pexip Distributed Gateway calls

The same principles described above also apply to calls placed via the Pexip Distributed Gateway:

- The call signaling will always be handled by the Conferencing Nodes that receive and place the call (you can configure Call Routing Rules to call out from a Conferencing Node in a different location to that which received the call).
- The call media will be handled by the Conferencing Node that has the most available capacity in that location.
- If there is no available capacity in that location and one or more overflow locations have been configured that do have available capacity, the media will be handled by one of the Conferencing Nodes in an overflow location.

Customizing the Pexip Infinity user experience

You can easily apply your own corporate branding to the Pexip Infinity platform, and produce a personalized user experience for all of your Pexip Infinity services.

This topic gives some examples of how themes (which are applied to one or more services, and affect all participants using those services) can be used to customize the voice prompts and images used in your Pexip Infinity services, and how branding customizations can be applied to change the look and feel of the Infinity Connect clients (Pexip's free video clients, which can optionally be used to access those services).

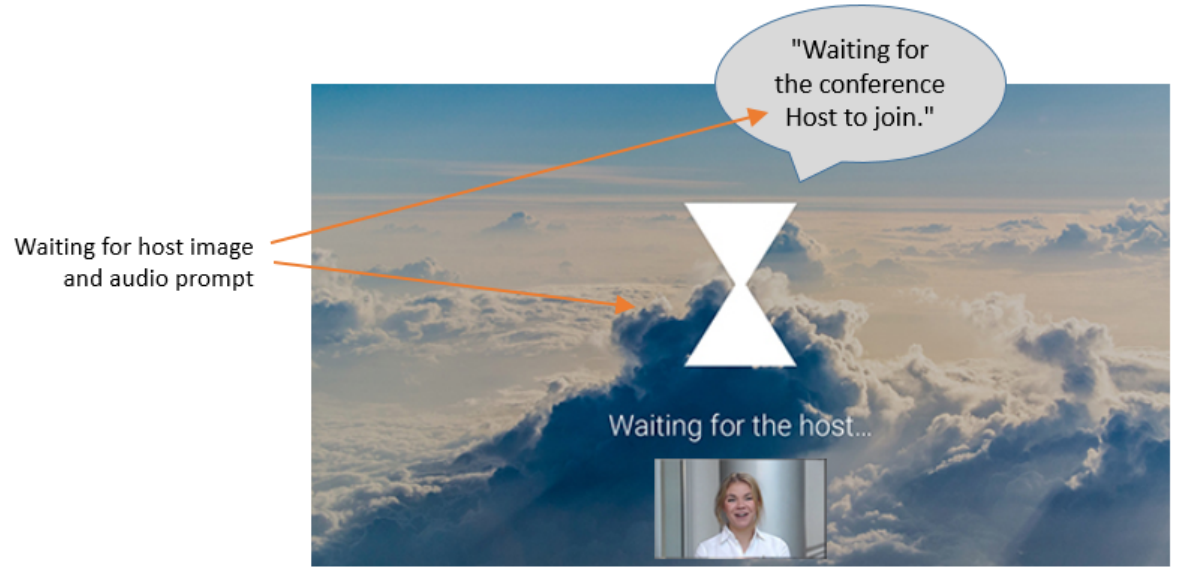
Themes

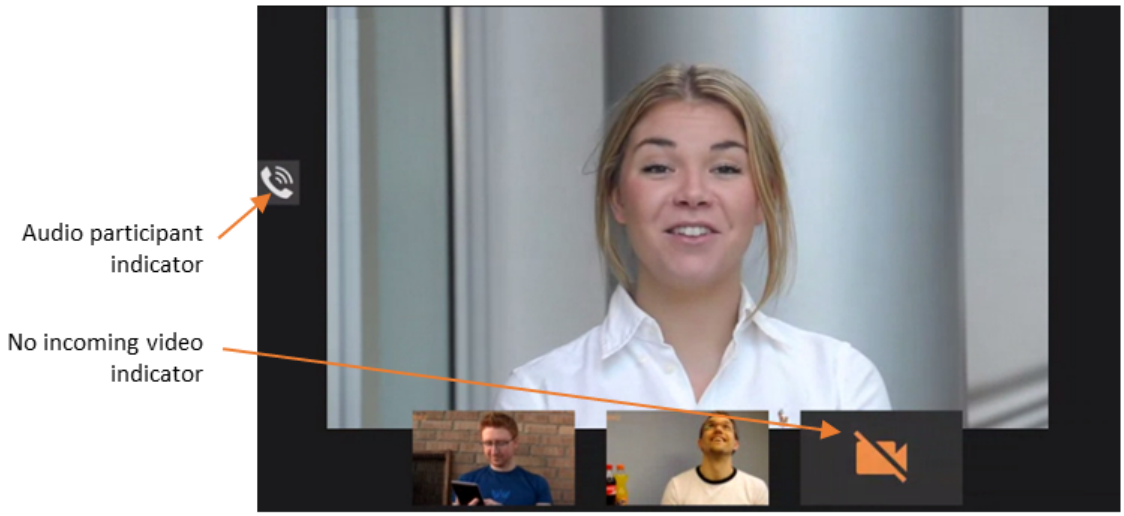
Themes are applied to the service itself (VMR, Virtual Auditorium or Virtual Reception) or Pexip Distributed Gateway call (via Call Routing Rules), and affect all participants using that service, regardless of the device or video client that the participant is using to make the call. A different theme can be applied to each service or rule if required.

Themes are used to control:

- all of the voice prompts used when accessing or participating in a conference
- the appearance of PIN entry screens (on devices other than the Infinity Connect clients)
- in-conference indicators such as audio-only participants, streaming, no incoming video etc.
- the appearance of information screens such as waiting for host, capacity exceeded, insufficient licenses etc.

The following images show some examples of which elements of the user experience can be customized via themes. All of the examples show a video client that is accessing a service that uses the base (default) Pexip theme.





Themes are uploaded and managed via the Pexip Infinity Administrator interface (**Service Configuration > Themes**).

For more information, including the full set of theme elements that can be customized, see Customizing images and voice prompts using themes.

Infinity Connect customization

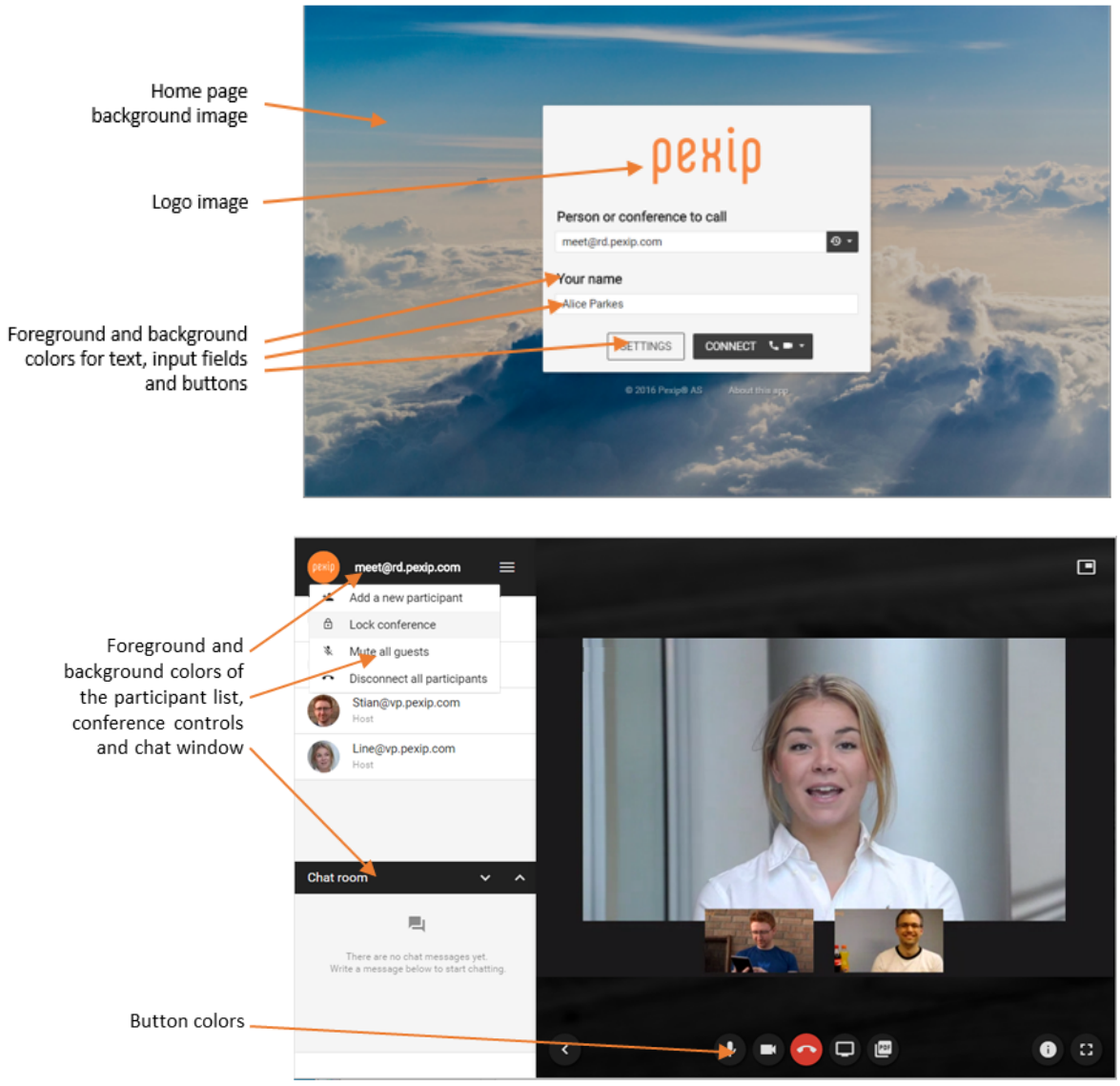
The branding and styling of the Infinity Connect Web App and the Infinity Connect desktop client can also be customized.

The participant has the same branding experience when using the application, regardless of which service is being accessed. (However, the theme-based elements of each service may also have been customized as described above.)

Infinity Connect customization can be used to control:

- the logos and background image displayed on the application's home page
- the foreground and background colors used for:
  - headings, input fields and buttons on the home page and other dialogs (such as PIN entry or adding a participant)
  - the participants list and chat window
  - in-call control buttons and the conference control menu options

The following images show examples of the Infinity Connect Web App using the default Pexip branding, and identifies the elements of the application that can be customized:



You can customize the Infinity Connect Web App by creating and then uploading a branding package to the Management Node (**Service Configuration > Web App Customization**).

There are two main ways to create a branding package for the Infinity Connect Web App:

- via the Pexip branding portal (https://branding.pexip.com)
- manual configuration of the default branding files after downloading them from the Management Node

For more information, see Customizing the Infinity Connect Web App.

If you also use the Infinity Connect desktop client, you can use the same set of customized files in both the Infinity Connect Web App and the Infinity Connect desktop client. See Customizing the Infinity Connect desktop client for more information.

Pexip Infinity installation guidelines

Pexip Infinity is installed as a Management Node and one or more Conferencing Nodes. All are virtual machines (VMs) that are either deployed onto host hardware using hypervisors, or hosted on a cloud platform service. The Management Node is deployed first, and is then used to configure the Pexip Infinity platform and deploy Conferencing Nodes.

For information about how to install and deploy Pexip Infinity, see:

Installation overview 47

Planning and prerequisites 47

Next steps after initial installation 70

Installation overview

Pexip Infinity is installed as a Management Node and one or more Conferencing Nodes. All are virtual machines (VMs) that are either deployed onto host hardware using hypervisors, or hosted on a cloud platform service. The Management Node is deployed first, and is then used to configure the Pexip Infinity platform and deploy Conferencing Nodes.

Full instructions on how to install Pexip Infinity, including how to install the Management Node, are available on our website and to download as individual hypervisor-specific installation guide PDFs.

Planning and prerequisites

Deciding on a deployment environment

Pexip Infinity supports a wide range of network scenarios, from small, local, private deployments to large, global, cloud-based implementations. How many Conferencing Nodes you deploy, where they are located, and the hypervisors you use will depend on the makeup and requirements of your individual organization.

While your Pexip authorized support representative is the best person to advise you on the most appropriate deployment environment, below are some of the factors to consider.

On-prem or cloud?

On-premise deployments use your own network, hardware and hypervisors. You will need to source appropriate servers to host the Management Node and Conferencing Nodes (for more information see Server design recommendations. You will also need to consider whether the majority of users are connecting from within your private network or whether you will have users connecting externally, and therefore whether your Conferencing Nodes need to be publicly accessible. If you already have a mature IT network with datacenters in appropriate locations, this may be a suitable option.

Cloud deployments use Amazon Web Services (AWS) or Microsoft Azure to host some or all of the Conferencing Nodes. This option does not require initial investment in hardware, but you will instead pay for usage.

You can also deploy Pexip Infinity using a **combination of on-prem and cloud**, with the Management Node and some Conferencing Nodes deployed within your own network, and other Conferencing Nodes deployed using a cloud service, for increased flexibility. You can even set up Dynamic bursting to the AWS cloud to give you temporary increases in capacity.

For more details on the on-prem and cloud options, see Network deployment options.

Hypervisor

For on-prem deployments, you will need to install a hypervisor on your servers before you can deploy the Management Node and Conferencing Nodes.

Pexip Infinity currently supports four hypervisors:

- **VMware** is an independent product with a range of features, dependent on the licensing options you chose.
- **Microsoft Hyper-V** is included with Windows Server and also offers a range of features.
- **KVM** and **Xen** are open-source options that are not as feature-rich.

For more information, see Supported hypervisors.

Number and location of Conferencing Nodes

The capacity of your Pexip Infinity deployment, in terms of how many calls can be handled at any one time, is largely determined by the number of Conferencing Nodes you deploy. To optimize the conferencing experience, we recommend that, where possible, you locate Conferencing Nodes close to each of your main user bases. For example, if the majority of your users are located in three main offices then you should consider concentrating the majority of your Conferencing Nodes among those locations. If however the majority of your users are spread across multiple geographic locations, if possible you should deploy Conferencing Nodes evenly among your datacenters.

For more information, see Capacity planning.

Server and network requirements

Host servers

The Management Node and Conferencing Nodes are virtual machines (VMs) that run on industry-standard host servers. A Management Node can run on the same host server as a Conferencing Node. Other Conferencing Nodes can run on host servers in the same or different locations, allowing you to create a globally distributed system.

We provide a guide to help you chose and configure appropriate servers on which to host your Management Node and Conferencing Node. For further information, see the **Pexip Infinity Server Design Guide**.

Network requirements

Depending on your network deployment scenario, you may have to configure your system to operate behind a static NAT and to ensure that traffic can be routed between nodes. See Network deployment options and Network routing and addressing options for Conferencing Nodes for more information about the requirements and implications of your deployment scenario.

Note that in all Pexip Infinity deployment scenarios:

- All Pexip nodes must be fully routable to each other (full mesh) in both directions. This means that the Management Node must be able to reach every Conferencing Node, and each Conferencing Node must be able to reach every other Conferencing Node.
- Any internal firewalls must be configured to allow UDP port 500 and traffic using IP protocol 50 (ESP) in both directions between all Pexip nodes.
- There cannot be a NAT between any Pexip nodes.

Call control

If your deployment includes a call control system, it must be configured to route calls to Pexip Infinity appropriately. The exact configuration will depend on your deployment and dial plan, but in general calls placed from an endpoint to a Virtual Meeting Room alias should be routed to the endpoint's local Conferencing Nodes. For more information, see Implementing a dial plan.

Lync / Skype for Business server

If your deployment includes Lync / Skype for Business, you must set up static routes to domains used by Pexip Infinity aliases. For more information, see **Pexip Infinity and Microsoft Lync / Skype for Business Deployment Guide**.

Supported hypervisors

Pexip Infinity currently offers specific support for VMware ESXi, Microsoft Hyper-V, KVM and Xen hypervisors. Other hypervisors and orchestration layers may be used but are not officially supported. If you wish to deploy Pexip Infinity using a non-supported hypervisor, we recommend that you contact your Pexip authorized support representative for assistance.

Note that suspending and resuming Virtual Machines that are running the Pexip Infinity Management Node or Conferencing Nodes is not supported.

The Pexip Infinity platform can also be deployed as a cloud service via **Amazon Web Services (AWS)** or **Microsoft Azure**, with private, public or hybrid deployment options.

VMware vSphere ESXi

VMware versions

Version 14.1 of the Pexip Infinity platform supports VMware vSphere ESXi 5.x and 6.0, although we recommend ESXi 5.5 or 6.0. Support for ESXi 4.1 is being deprecated - if you have upgraded from a version prior to v12, you can still deploy Conferencing Nodes to servers running ESXi 4.1; however if you have a new deployment using v12 or later and attempt to deploy a Conferencing Node to a server running ESXi 4.1, that node will go straight into maintenance mode.

You must have a suitable VMware environment already installed. For information on configuring a VMware environment specifically for use with Pexip Infinity, see Configuring VMware and Advanced VMware ESXi administration.

VMware editions

The Pexip Infinity platform will run on the **free edition** of vSphere Hypervisor. However, this edition has a number of limitations (limited support from VMware, no access to vCenter or vMotion, no access to VMware API). In particular, the lack of access to the VMware API means that all Conferencing Nodes will have to be deployed manually (see Deployment types). For this reason we do not recommend its use except in smaller deployments, or test or demo environments.

The minimum edition of VMware that we recommend is the vSphere **Standard edition**. This does not have the limitations of the free edition. If you do not already use VMware in your enterprise, the vSphere **Essentials Kit** is a simple way to get started and will provide you with Standard edition licenses for 3 servers (with 2 CPUs each) plus a vCenter license.

The **Enterprise Plus edition** includes further additional features relevant to the Pexip Infinity platform that could be of benefit to larger deployments. These include Storage DRS and Distributed Switch.

For a comparison of the VMware editions, see **http://www.vmware.com/products/vsphere.html#compare**.

Microsoft Hyper-V

Version 14.1 of Pexip Infinity supports Microsoft Hyper-V in the form of Microsoft Hyper-V Server 2012 and later, or Windows Server 2012 and later. You must have a suitable Hyper-V environment already installed. For information on configuring a Hyper-V environment specifically for use with Pexip Infinity, see Configuring Hyper-V.

KVM

For information on configuring a KVM environment specifically for use with Pexip Infinity, see **Configuring KVM for Pexip Infinity**.

Xen

Pexip Infinity requires Xen 4.2 and later. For information on configuring a Xen environment specifically for use with Pexip Infinity, see **Configuring Xen for Pexip Infinity**.

Other hypervisors and orchestration layers

Pexip Infinity Management Nodes and Conferencing Nodes support static IP addressing only. Therefore the hypervisor or orchestration layer must also support static IP addressing.

Cloud service providers

The Pexip Infinity platform can be deployed on Amazon Web Services (AWS) and Microsoft Azure cloud platforms.

See Deploying Pexip Infinity on Amazon Web Services and Deploying Pexip Infinity on Microsoft Azure for installation guidelines and prerequisites.

About the Pexip Infinity software files

The Pexip Infinity platform is entirely software-based, and is deployed by installing the files provided by Pexip onto your hypervisor or cloud platform to create Virtual Machines (VMs), which you then configure appropriately for your environment.

The Pexip Infinity version 14.1 release includes the files described below. Which set of files you require depends on:

- the hypervisor or cloud platform you are deploying on
- whether you are deploying a new instance of Pexip Infinity, upgrading an existing deployment, or deploying a new Conferencing Node
- if you are deploying new generic Conferencing Nodes in unsupported or non-standard environments.

Upgrading an existing deployment (all environments)

The Pexip Infinity upgrade package is required when upgrading an existing deployment. It is used to upgrade all deployment environments.

Description	File name	Location	Hypervisor	Notes
Pexip Infinity upgrade package	Pexip_Infinity_v14.1_UPGRADE_<build>.tar	www.pexip.com/software-download	Any hypervisor or cloud platform	Used when **upgrading** the Pexip Infinity platform from version 8 or later to version 14.1. For more information, see Upgrading the Pexip Infinity platform.

Deploying the Management Node on an on-prem hypervisor

Use one of these files when deploying the Management Node on an on-prem hypervisor.

Description	File name	Location	Hypervisor	Notes
Pexip Infinity OVA	Pexip_Infinity_v14.1_generic_pxMgr_<build>.ova	www.pexip.com/software-download	Any hypervisor **except** Microsoft Hyper-V	Used to deploy a new generic **Management Node** VM using VMware, KVM, Xen or other hypervisors except Microsoft Hyper-V. For more information, see: • Installing the Management Node - VMware • Installing the Management Node - KVM • Installing the Management Node - Xen • Installing the Management Node VM — other hypervisors
Pexip Infinity ZIP	Pexip_Infinity_v14.1_HyperV_pxMgr_<build>.zip	www.pexip.com/software-download	Microsoft Hyper-V	Used to deploy a generic **Management Node** VM using Hyper-V. For more information, see Installing the Management Node VM.

Manually deploying new Conferencing Nodes on supported hypervisors

There is no requirement to download any files from the Pexip website when manually deploying new Conferencing Nodes on VMware, Hyper-V, KVM or Xen hypervisors, as the relevant file is generated on request by the Management Node.

- For VMware, KVM and Xen, a file in the format **pexip-<hostname>.<domain>.ova** is generated. See Deploying new Conferencing Nodes for full information.
- For Hyper-V, a file in the format **pexip-<hostname>.<domain>.zip** is generated. See Manually deploying a Conferencing Node on a Hyper-V host for full information.

Automatically deploying new Conferencing Nodes on VMware

You can automatically deploy a new Conferencing Node on to a VMware ESXi host.

When deploying a Conferencing Node automatically, the Pexip Infinity Management Node generates a file that contains a template for a generic VM along with Conferencing Node-specific configuration details, and uploads it automatically onto the hypervisor. The new Conferencing Node VM is then created without any further intervention by the administrator.

There is no requirement to download any files from the Pexip website when automatically deploying new Conferencing Nodes.

For more information, see Automatically deploying a new Conferencing Node on a VMware host.

Deploying the Management Node or Conferencing Nodes on cloud environments (AWS and Azure)

There is no requirement to download any files from the Pexip website when you are deploying the Pexip Infinity Management Node or Conferencing Nodes on a cloud platform — the relevant Pexip image files are hosted on the AWS/Azure platform instead.

See Deploying Pexip Infinity on Amazon Web Services and Deploying Pexip Infinity on Microsoft Azure for more information.

Manually deploying new generic Conferencing Nodes on non-supported hypervisors or orchestration layers

Use one of these files when deploying a Conferencing Node on other non-supported hypervisors or orchestration layers.

Description	File name	Location	Hypervisor	Notes
Generic Pexip Infinity Conferencing Node OVA	Pexip_Infinity_v14.1_generic_ConfNode_<build>.ova	https://dl.pexip.com/infinity/index.html and then select the appropriate directory for your software version	Other environments that take .ova or .ovf files as input	Used to manually deploy a generic **Conferencing Node** VM in environments that take .ova or .ovf files as input. For more information, see Deploying a Conferencing Node using a generic VM template and configuration file.
Generic Pexip Infinity Conferencing Node ZIP	Pexip_Infinity_v14.1_HyperV_ConfNode_<build>.zip	https://dl.pexip.com/infinity/index.html and then select the appropriate directory for your software version	Hyper-V in a cloud-based environment	Used to manually deploy a generic **Conferencing Node** VM using Hyper-V in cloud-based environments or other orchestration layers where standard manual deployment is problematic. For more information, see Deploying a Conferencing Node using a generic VM template and configuration file.

Network deployment options

If you need to support business-to-business video calls and provide access to Pexip Infinity resources from external systems and endpoints such as remote or federated Lync / Skype for Business clients, remote SIP and H.323 endpoints, and Infinity Connect clients, you need to consider how to deploy your Pexip Infinity Conferencing Nodes.

This section explains how the Pexip Infinity platform fits into typical network deployment scenarios:

- a private "on-premises" deployment of privately-addressed Pexip Infinity nodes
- a public DMZ deployment with publicly-addressable Pexip Infinity nodes (with optional static NAT)
- a combination of public DMZ and "on-premises" nodes
- deploying Conferencing Nodes with dual network interfaces (one "internal" interface for inter-node communication, and one "external" interface for signaling and media to endpoints and other video devices)

The Pexip Infinity platform can also be deployed as a cloud service via **Amazon Web Services (AWS)** or **Microsoft Azure**, with private, public or hybrid deployment options.

General network requirements

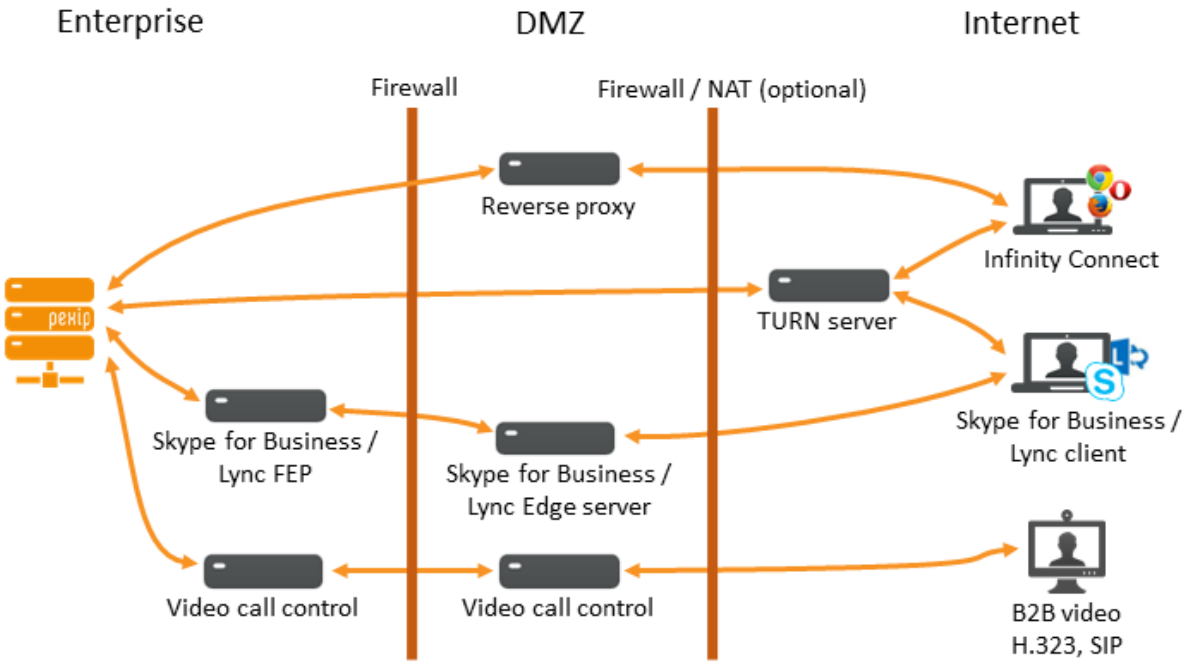
Note that in all Pexip Infinity deployment scenarios:

- All Pexip nodes must be fully routable to each other (full mesh) in both directions. This means that the Management Node must be able to reach every Conferencing Node, and each Conferencing Node must be able to reach every other Conferencing Node.
- Any internal firewalls must be configured to allow UDP port 500 and traffic using IP protocol 50 (ESP) in both directions between all Pexip nodes.
- There cannot be a NAT between any Pexip nodes.

For information about using the Pexip Reverse Proxy and TURN Server to provide access to privately-addressed Pexip Infinity nodes from clients situated on the public internet, and for integration details with other third-party call management systems, see Integrating Pexip Infinity with other systems.

Privately-addressed "on-premises" nodes

A typical "on-premises" deployment of privately-addressed Pexip Infinity nodes is shown below:

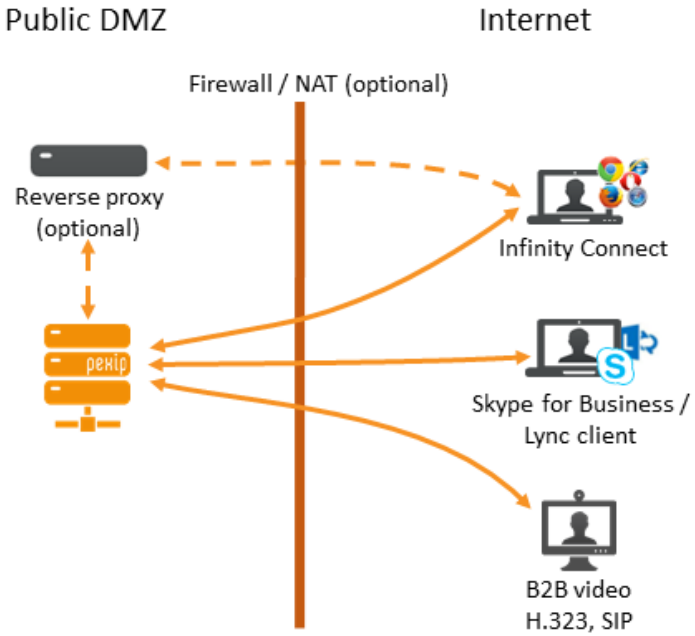


In this type of deployment:

- The Management Node and the Conferencing Nodes are deployed with private IP addresses in the local enterprise network.
- External SIP and H.323 endpoints and other forms of business-to-business video calls connect via a firewall traversal / video call control solution such as a Cisco VCS.
- External Lync / Skype for Business clients connect to Pexip Infinity via a Lync / Skype for Business Edge server and media is routed through a TURN server. For Lync / Skype for Business communications, a separate STUN server is required if the TURN server is behind the same NAT device as your Conferencing Nodes.
- Infinity Connect WebRTC clients connect via a reverse proxy and route their media through a TURN server. (Note that RTMP clients cannot establish audio/video media paths with Pexip Infinity via a TURN server.)
- The external firewall can optionally be a NAT device.

Publicly-addressed DMZ nodes

A typical deployment of publicly-routable Pexip Infinity nodes is shown below:



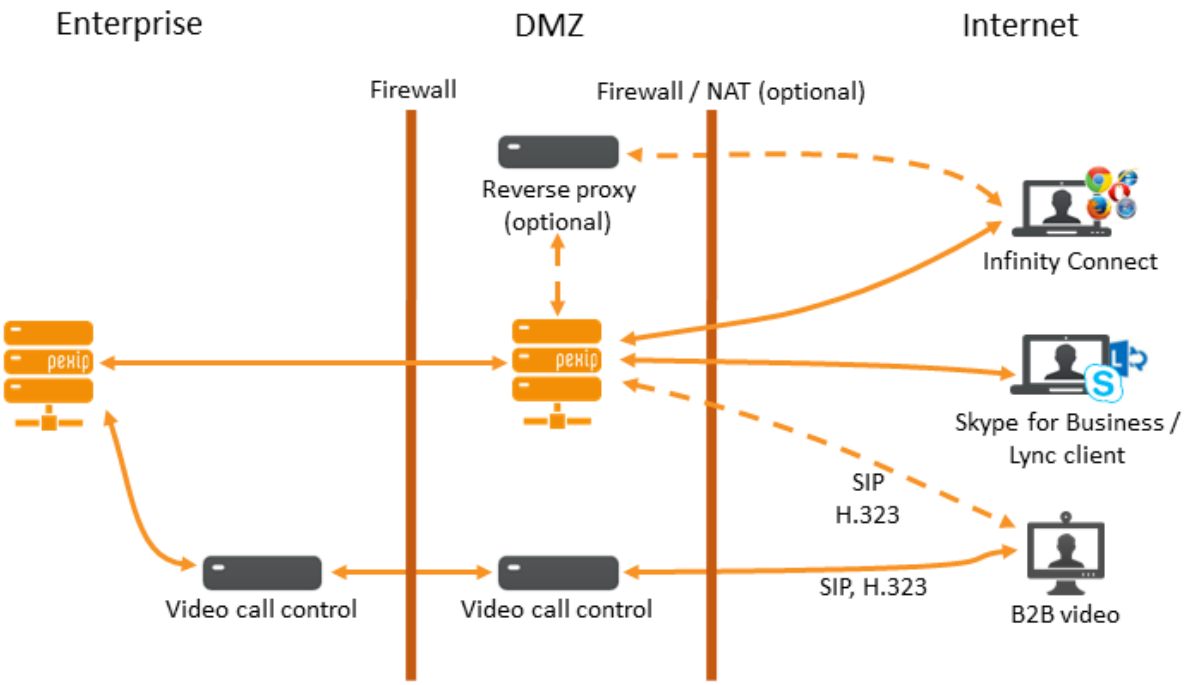
In this type of deployment:

- One or more Conferencing Nodes are deployed in a public DMZ. They have publicly-reachable IP addresses — either directly or via static NAT.
- External SIP and H.323 endpoints and other forms of business-to-business video calls can connect directly to the public-facing IP addresses of the Conferencing Nodes.
- External Lync / Skype for Business clients connect to Pexip Infinity and media can be routed directly through the public-facing IP addresses of the Conferencing Nodes (an on-premises Lync / Skype for Business server is not needed). As Conferencing Nodes support ICE, a TURN server is not required.
- Infinity Connect clients can connect to Pexip Infinity and media can be routed directly to the Conferencing Nodes. You can optionally deploy a reverse proxy for load balancing or branding.
- Note that the Management Node will typically be deployed with a private IP address in the local enterprise network. You must ensure that there is no NAT between the Management Node and the Conferencing Nodes in the DMZ.

Combining public DMZ and privately-addressed nodes

Depending on your requirements, aspects of the privately-addressed and public DMZ scenarios can be combined. For example, as shown below, you may want to:

- deploy some publicly-accessible Pexip Infinity nodes for external Lync / Skype for Business and Infinity Connect client access
- deploy some privately-accessible nodes and route external SIP and H.323 endpoints and other forms of business-to-business video calls to them via a video call control solution.



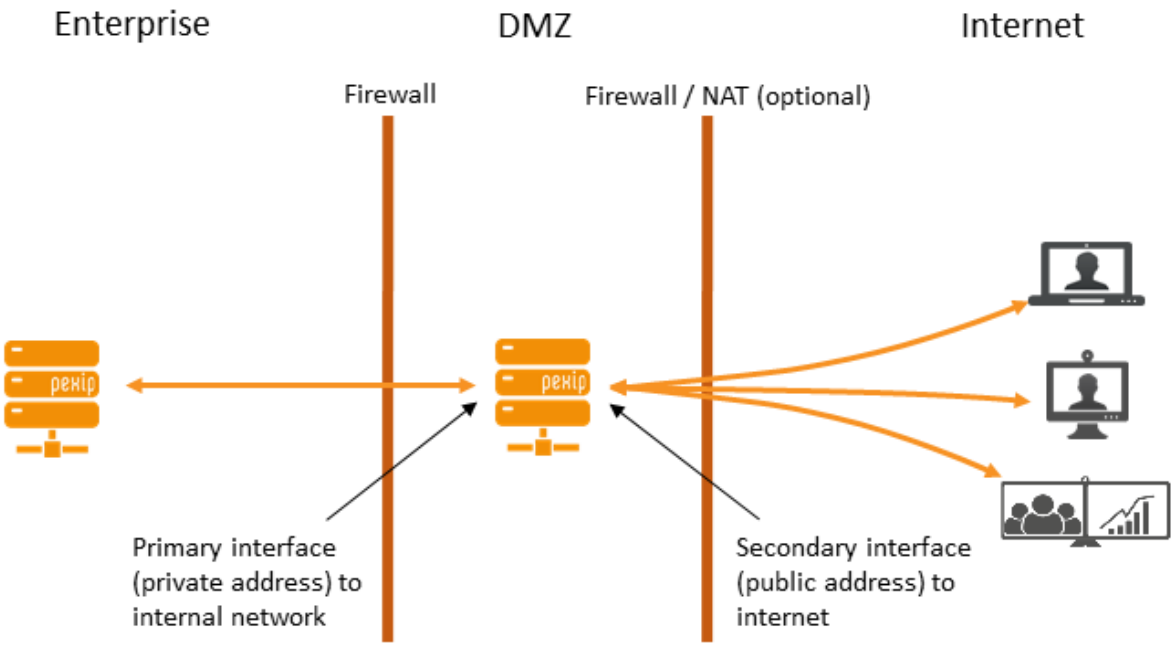
In this example deployment:

- One or more Conferencing Nodes are deployed in a public DMZ. They have publicly-reachable IP addresses — either directly or via static NAT.
- Additional Conferencing Nodes are deployed with private IP addresses in the local enterprise network. They must be assigned to a different system location than the publicly-addressable nodes.
- External SIP and H.323 endpoints and other forms of business-to-business video calls are routed via a video call control solution such as a Cisco VCS. Alternatively, SIP and H.323 endpoints can register to Conferencing Nodes in the DMZ and route their calls through those nodes.
- External Lync / Skype for Business clients connect to Pexip Infinity and media can be routed directly through the public-facing IP addresses of the Conferencing Nodes (an on-premises Lync / Skype for Business server is not needed). As Conferencing Nodes support ICE, a TURN server is also not required.
- Infinity Connect clients can connect to Pexip Infinity and media can be routed directly to the Conferencing Nodes. You can optionally still deploy a reverse proxy for load balancing or branding.
- Geo backplanes are established between the privately-addressed nodes and the publicly-addressed nodes as and when required. Note that there cannot be a NAT between Pexip Infinity nodes (any nodes in the local enterprise network and the nodes in the DMZ).
- Static routing configuration may be required to allow those nodes in the public DMZ to communicate with Pexip Infinity nodes or other systems in the local enterprise network.

Conferencing Nodes with dual network interfaces (NICs)

For additional deployment flexibility, you can configure a secondary network address on a Conferencing Node.

You would



typically deploy a Conferencing Node with dual network interfaces when it is connected to a dedicated video zone or it is being deployed in a public DMZ where the primary interface would be to an internal, private network segment within the enterprise (where it can connect to the Management Node and other Conferencing Nodes) and the secondary interface would be towards the video zone or on the publicly-addressable side of the DMZ perimeter network and used for connecting to external endpoints and devices.

When a secondary network address is configured:

- The primary address is always used for inter-node communication to the Management Node and to other Conferencing Nodes.
- The secondary address is always used for signaling and media (to endpoints and other video devices).
- You can have a mixture of any number of single-interfaced and dual-interfaced Conferencing Nodes, providing all nodes can communicate with each other via their primary interfaces.

Note that dual network interfaces are not supported on Conferencing Nodes deployed in public cloud services (AWS and Azure).

For more information on configuring dual network interfaces, see Deploying Conferencing Nodes with dual network interfaces (NICs) and Example deployment scenarios.

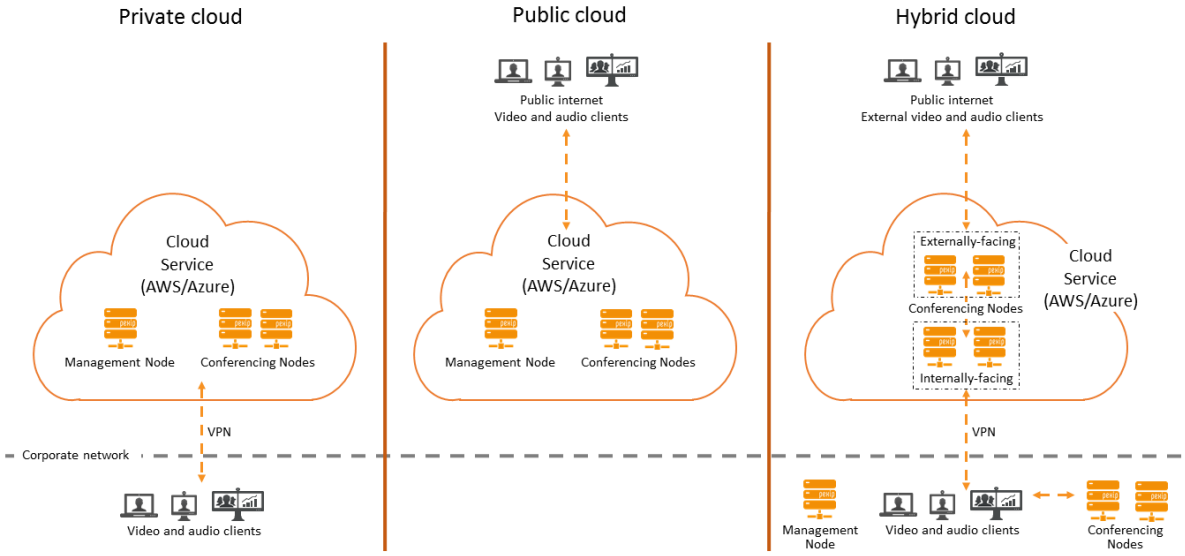
Deploying as a cloud service via Amazon Web Services or Microsoft Azure

The Pexip Infinity platform can be deployed in the Amazon Web Services (AWS) or Microsoft Azure cloud.

There are three main deployment options for your Pexip Infinity platform when using a cloud service:

- **Private cloud**: all nodes are deployed within a private cloud service. Private addressing is used for all nodes and connectivity is achieved by configuring a VPN tunnel from the corporate network to the cloud network. As all nodes are private, this is equivalent to an on-premises deployment which is only available to users internal to the organization.
- **Public cloud**: all nodes are deployed within the cloud network. All nodes have a private address but, in addition, public IP addresses are allocated to each node. The node's private addresses are only used for inter-node communications. Each node's public address is then configured on the relevant node as a static NAT address. Access to the nodes is permitted from the public internet, or a restricted subset of networks, as required. Any systems or endpoints that will send signaling and media traffic to those Pexip Infinity nodes must send that traffic to the public address of those nodes. If you have internal systems or endpoints communicating with those nodes then you must ensure that your local network allows this.

- **Hybrid cloud**: the Management Node, and optionally some Conferencing Nodes, are deployed in the corporate network. A VPN tunnel is created from the corporate network to the cloud network. Additional Conferencing Nodes are deployed in the cloud network and are managed from the on-premises Management Node. The cloud-hosted Conferencing Nodes can be either internally-facing, privately-addressed (private cloud) nodes; or externally-facing, publicly-addressed (public cloud) nodes; or a combination of private and public nodes (where the private nodes are in a different Pexip Infinity system location to the public nodes).



See Deploying Pexip Infinity on Amazon Web Services and Deploying Pexip Infinity on Microsoft Azure for more information.

Example deployment scenarios

Here are some example Pexip Infinity deployment scenarios that demonstrate the various network deployment options that are available when deploying the Pexip Infinity platform:

- Public DMZ node with single NIC and externally-facing default gateway (with/without NAT)
- Public DMZ node with single NIC and three-legged firewall (with/without NAT)
- Public DMZ node with dual NICs and externally-facing default gateway (with/without NAT)
- Conferencing Node with dual NICs and routing to a video zone (with/without NAT)

For more information about how to configure static NAT, dual network interfaces and static routes, see Network routing and addressing options for Conferencing Nodes.

Public DMZ node with single NIC and externally-facing default gateway (with/without NAT)

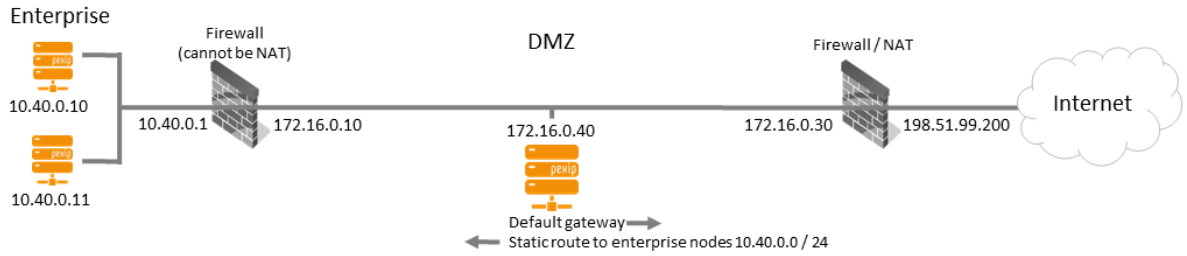
These types of deployment have a Conferencing Node in the public DMZ with:

- a single externally-facing interface (default gateway is out to the public internet) — this can be with or without NAT
- a static route back into the enterprise (to communicate with any private Conferencing Nodes and the Management Node)
- an internal firewall (which must not be NAT) between the enterprise nodes and the public DMZ nodes.

The static route must be applied to the Conferencing Node during its initial deployment phase (otherwise it will not be able to communicate with the Management Node and pick up its configuration).

Externally-facing interface with NAT

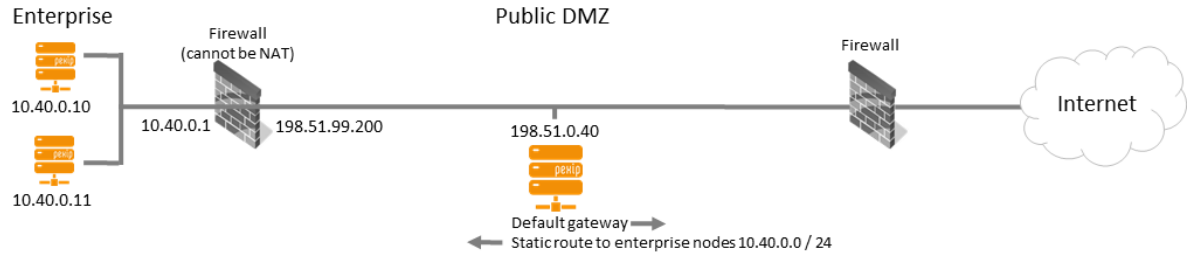
In this example the external firewall performs NAT:



DMZ node has its default gateway at 172.16.0.30 (external firewall)
DMZ node has static route* to 10.40.0.0 / 24 with gateway IP address of 172.16.0.10
* Ensure that the static route is applied to the Conferencing Node during its initial deployment phase

Externally-facing interface without NAT

In this example the external firewall does not perform NAT:



Public DMZ node with its default gateway towards public internet
Public DMZ node has static route* to 10.40.0.0 / 24 with gateway IP address of 198.51.99.200
* Ensure that the static route is applied to the Conferencing Node during its initial deployment phase

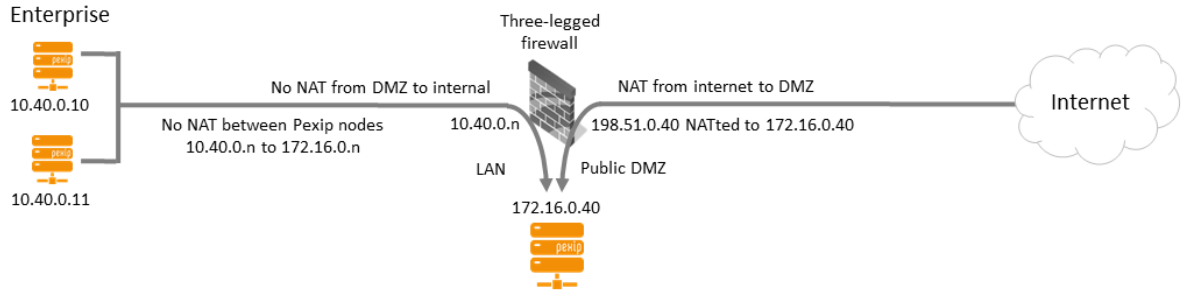
Public DMZ node with single NIC and three-legged firewall (with/without NAT)

These types of deployment have a Conferencing Node in the public DMZ with:

- a single interface — this can be with or without NAT for traffic that is destined for the public internet
- a three-legged firewall that allows bi-directional routing between enterprise nodes in the LAN (any private Conferencing Nodes and the Management Node) — this must always be without NAT.

NAT applied to external traffic

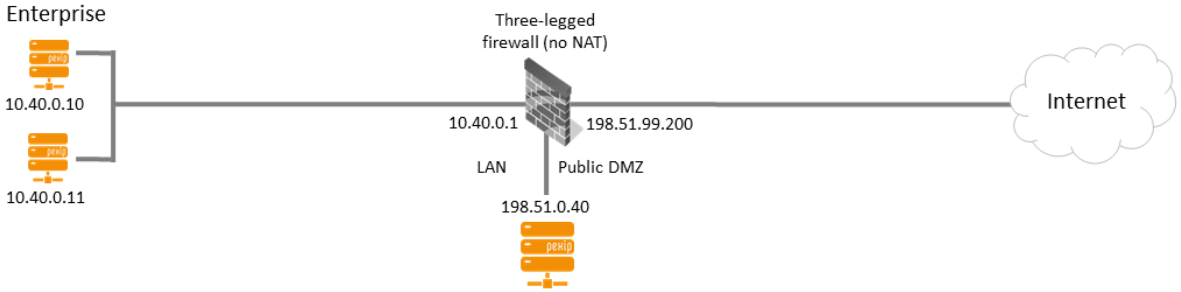
In this example the firewall applies NAT to external traffic (between DMZ and internet):



Firewall allows bi-directional routing between LAN enterprise nodes and public DMZ node. Traffic between DMZ node and internet is NATted.

NAT is not applied to external traffic

In this example the firewall does not apply NAT to external traffic (between DMZ and internet):



Firewall allows bi-directional routing between LAN enterprise nodes and public DMZ node. Traffic between DMZ node and internet is not NATted.

Public DMZ node with dual NICs and externally-facing default gateway (with/without NAT)

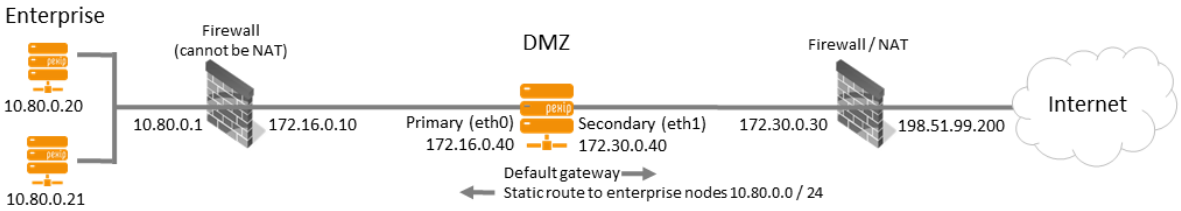
These types of deployment have a Conferencing Node in the public DMZ with:

- dual network interfaces:
  - the primary NIC is internally-facing (eth0 must be used for all inter-node communication)
  - the secondary NIC is externally-facing (eth1 must be used for signaling and media to endpoints and other video devices) — this can be with or without NAT
- the default gateway is out to the public internet (and thus is associated automatically with the secondary NIC)
- a static route back into the enterprise (to communicate with any private Conferencing Nodes and the Management Node)
- an internal firewall (which must not be NAT) between the enterprise nodes and the public DMZ nodes.

The static route must be applied to the Conferencing Node during its initial deployment phase (otherwise it will not be able to communicate with the Management Node and pick up its configuration).

Externally-facing interface with NAT

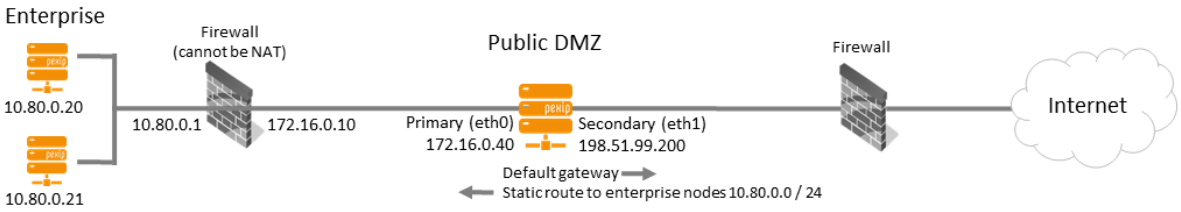
In this example the external firewall performs NAT:



DMZ node has its default gateway at 172.30.0.30 (external firewall) and is associated with secondary interface (eth1)
DMZ node has static route* to 10.80.0.0 / 24 with gateway IP address of 172.16.0.10
* Ensure that the static route is applied to the Conferencing Node during its initial deployment phase

Externally-facing interface without NAT

In this example the external firewall does not perform NAT:



Public DMZ node with its default gateway towards public internet
Public DMZ node has static route* to 10.80.0.0 / 24 with gateway IP address of 172.16.0.10
* Ensure that the static route is applied to the Conferencing Node during its initial deployment phase

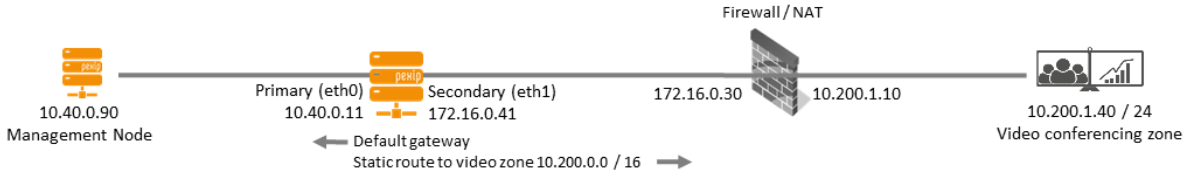
Conferencing Node with dual NICs and routing to a video zone (with/without NAT)

These types of deployment have a Conferencing Node that communicates with a dedicated video conference VLAN. It has:

- dual network interfaces:
  - the primary NIC is internally-facing (eth0 must be used for all inter-node communication)
  - the secondary NIC faces the video zone (eth1 must be used for signaling and media to endpoints and other video devices in the video zone) — this can be with or without NAT
- the default gateway is to the Pexip nodes (and thus is associated automatically with the primary NIC)
- a static route for traffic destined for the video zone.

Video zone routing with NAT

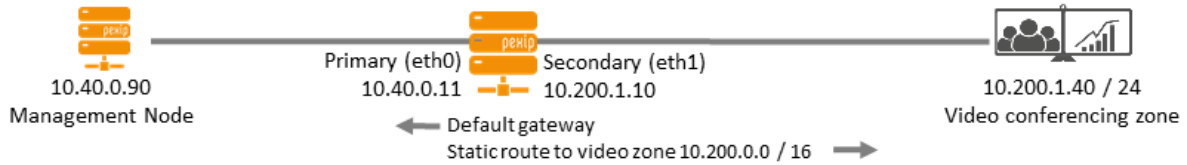
In this example the traffic sent to the video zone has NAT applied:



Conferencing Node default gateway associated with primary interface (eth0) towards Management Node
Conferencing Node static route to 10.200.0.0 / 16 video zone subnet with gateway IP address of 172.16.0.30

Video zone routing without NAT

In this example the traffic sent to the video zone does not have NAT applied:



Conferencing Node default gateway associated with primary interface (eth0) towards Management Node
Conferencing Node static route to 10.200.0.0 / 16 video zone subnet

Network routing and addressing options for Conferencing Nodes

This section describes Pexip Infinity's network routing and addressing configuration options that are often required when deploying Conferencing Nodes in a public DMZ, or when routing to a dedicated video zone. It covers the following scenarios:

- Configuring Pexip Infinity nodes to work behind a static NAT device
- Applying static routes to enable routing between externally-facing nodes and local network nodes
- Deploying Conferencing Nodes with dual network interfaces (NICs)
- Remote SIP endpoints behind a remote firewall/NAT

Note that in all Pexip Infinity deployment scenarios:

- All Pexip nodes must be fully routable to each other (full mesh) in both directions. This means that the Management Node must be able to reach every Conferencing Node, and each Conferencing Node must be able to reach every other Conferencing Node.
- Any internal firewalls must be configured to allow UDP port 500 and traffic using IP protocol 50 (ESP) in both directions between all Pexip nodes.
- There cannot be a NAT between any Pexip nodes.

In addition, you must ensure that all appropriate firewall ports have been opened as described in Pexip Infinity port usage and firewall guidance.

Further information is available in Network deployment options and Example deployment scenarios.

Configuring Pexip Infinity nodes to work behind a static NAT device

To configure your Pexip Infinity deployment to work behind a static NAT device (from the perspective of clients located on the Internet or in a dedicated video zone) you must:

1. Configure the NAT device / firewall with the static, publicly-reachable IP address of each Conferencing Node that you want to be accessible from devices in the internet / video zone, and then map the public address to the node's corresponding internal IP address. Note that it must be a 1:1 NAT.
2. Configure each publicly-reachable Conferencing Node with its **IPv4 static NAT address** (**Platform Configuration > Conferencing Nodes**) i.e. the public address of the node that you have configured on the NAT device.

Note that:

- Any Conferencing Nodes that are configured with a static NAT address must not be configured with the same **System location** as nodes that do not have static NAT enabled. This is to ensure that load balancing is not performed across nodes servicing external clients and nodes that can only service private IP addresses.
- Static NAT must be on the secondary interface if the Conferencing Node has dual network interfaces.
- Any internal systems such as Cisco VCSs or endpoints that will send signaling and media traffic to Pexip Infinity nodes that are enabled for static NAT should send that traffic to the public address of those nodes. You must ensure that your local network allows this.
- We do not recommend that you allow the Management Node to be accessible from devices in the public internet. However, if you want to do this, you must assign and configure the Management Node with its static NAT address. You should also configure your firewall to only allow access to the Management Node from the specific IP addresses from where you want to allow management tasks to be performed.
- There cannot be a NAT device between any Pexip Infinity nodes.

Deploying Conferencing Nodes with dual network interfaces (NICs)

For additional deployment flexibility, you can configure a secondary network address on a Conferencing Node.

You would typically deploy a Conferencing Node with dual network interfaces when it is connected to a dedicated video zone or it is being deployed in a public DMZ where the primary interface would be to an internal, private network segment within the enterprise (where it can connect to the Management Node and other Conferencing Nodes) and the secondary interface would be towards the video zone or on the publicly-addressable side of the DMZ perimeter network and used for connecting to external endpoints and devices.

As with specifying the IP address of a Conferencing Node with a single interface, the IP address and network mask of the secondary interface must be specified when the Conferencing Node is initially deployed and cannot be subsequently modified.

When a secondary network address is configured:

- The primary address is always used for inter-node communication to the Management Node and to other Conferencing Nodes.
- The secondary address is always used for signaling and media (to endpoints and other video devices).
- You can have a mixture of any number of single-interfaced and dual-interfaced Conferencing Nodes, providing all nodes can communicate with each other via their primary interfaces.
- The Conferencing Node's default gateway can be in either subnet and is automatically associated with the correct interface.
- Static routes can be in either subnet and are automatically associated with the correct interface.
- Static NAT, if configured, must be on the secondary interface.

Applying static routes to enable routing between externally-facing nodes and local network nodes

In some deployment scenarios you may have the Management Node and some Conferencing Nodes deployed in the local enterprise network, and some Conferencing Nodes deployed in a public DMZ or connected to a dedicated video zone.

If the default gateway on those non-local nodes is configured to route traffic out to the internet / video zone, you will need to configure static routes on those Conferencing Nodes to allow them to communicate with Pexip Infinity nodes or other systems in the local, internal network. See Managing static routes for information about to how to configure and assign static routes to Pexip Infinity nodes.

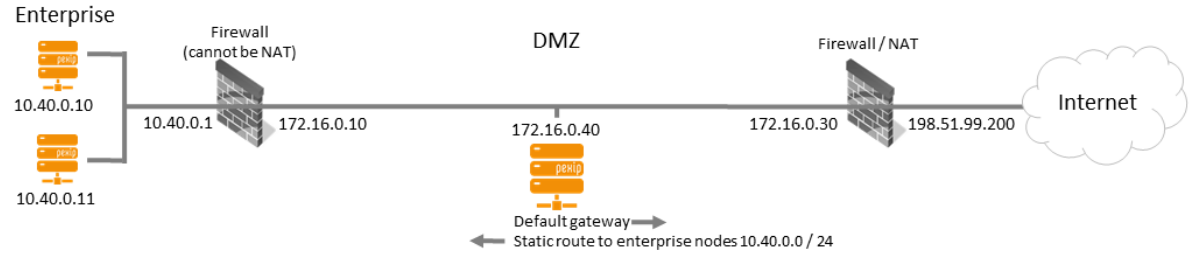
In situations where the Conferencing Node's default gateway is not towards the Management Node, the static route back to the Pexip Infinity platform must be applied to the Conferencing Node during its initial deployment phase (otherwise it will not be able to communicate with the Management Node and pick up its configuration).

Note that:

- Conferencing Nodes in a DMZ must not be configured with the same **System location** as nodes in a local network. This is to ensure that load balancing is not performed across nodes in the DMZ and nodes in the local network.
- You must configure any internal firewall to allow UDP port 500 and traffic using IP protocol 50 (ESP) in both directions between the Pexip Infinity nodes in the DMZ and the nodes in the local network.
- The firewall between the Pexip Infinity nodes in the DMZ and the nodes in the local network cannot be a NAT device.
- The external firewall between Conferencing Nodes in the DMZ and the internet can be configured with static NAT. In this case you would also need to configure each Conferencing Node in the DMZ with its relevant static NAT address.
- If you deploy the Management Node in the DMZ (although we do not recommend this for security reasons), it must also be assigned with the relevant static route (**Platform Configuration > Management Node**).

Example

For example, consider a Pexip Infinity system which is deployed as shown below:



DMZ node has its default gateway at 172.16.0.30 (external firewall)
DMZ node has static route* to 10.40.0.0 / 24 with gateway IP address of 172.16.0.10
* Ensure that the static route is applied to the Conferencing Node during its initial deployment phase

Example network with Pexip Infinity nodes in LAN and DMZ

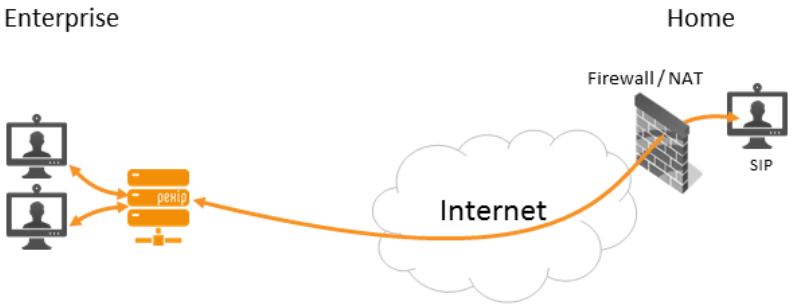
This deployment has:

- Pexip Infinity nodes on the enterprise LAN with addresses in the 10.40.0.0/24 subnet
- an internal firewall (without NAT) with LAN address 10.40.0.1 and DMZ address 172.16.0.10
- an external firewall with DMZ address 172.16.0.30 and public address 198.51.99.200
- a Conferencing Node in the DMZ with a single network interface with address 172.16.0.40 and which is configured with a default gateway address of 172.16.0.30 (the external firewall).

In this situation the Conferencing Node in the DMZ will, by default, send all of its traffic out through its default gateway — the external firewall at 172.16.0.30. To ensure that traffic from the Conferencing Node in the DMZ that is destined for Pexip Infinity nodes on the enterprise LAN can be routed to those nodes, you must deploy the Conferencing Node with a suitable static route applied to it. In this example the **Destination network address** of the static route would be **10.40.0.0** with a **Network prefix** of **24** (to route addresses in the range **10.40.0.0** to **10.40.0.255**) and the **Gateway IP address** would be 172.16.0.10 (the DMZ address of the internal firewall).

Remote SIP endpoints behind a remote firewall/NAT

Remote SIP endpoints that are behind remote firewalls/NATs can join Pexip Infinity conferences.



Media latching with remote endpoints behind a remote firewall/NAT

You do not have to apply any explicit configuration to Conferencing Nodes in order to allow remote SIP endpoints behind remote firewalls/NATs to join a conference. Pexip Infinity automatically uses signaling and media port latching to establish routable paths.

(Port latching involves Pexip Infinity waiting until it receives signaling and media traffic from the remote endpoint, and then it uses — or "latches" on to — the source address and port of that traffic as a destination for all traffic bound in the opposite direction. Typically these source addresses/ports will belong to the public interface of the NAT in front of the remote endpoint, and thus anything sent by Pexip Infinity to that address/port should ultimately reach the endpoint.)

Capacity planning

The capacity of each Conferencing Node in your Pexip Infinity deployment - in terms of the number of connections* that can be handled simultaneously - depends on a variety of factors including:

- server capacity and hardware configuration
- types of calls - Full HD, HD, SD, or audio-only, and whether there is a presentation stream included
- the number of unique VMRs being used (and thus the number of backplanes being reserved)
- the type of gateway call - if the call is distributed or not, and the types of client involved in the call
- the number of licenses that are available.

* A connection can be a call or presentation from an endpoint to a Virtual Meeting Room or Virtual Auditorium, a backplane between Conferencing Nodes, or a call into or out of the Pexip Distributed Gateway. In this context, a **connection** is analogous to a **port**.

The following sections explain each of these factors. For some comprehensive examples showing how these different factors can combine to influence capacity, see Resource allocation examples.

Server capacity

The capacity and configuration of the server on which the Conferencing Node is running will impact the number of calls that can be handled. This is influenced by a number of factors, including:

- processor generation
- number of cores
- processor speed
- hypervisor
- BIOS settings

For more information, see Server design recommendations and Example Conferencing Node server configurations.

Types of call

The type of call (HD, audio-only and so on) affects the amount of resource required to handle the call.

A single HD call requires roughly:

- half the resources of a Full HD call
- twice the resources of an SD call, or
- 12 times the resources of an audio-only call.

On startup, each Conferencing Node runs an internal capacity check. This capacity is translated into an estimated maximum number of HD, SD or audio-only calls, and can be viewed on the status page (**Status > Conferencing Nodes**) for each Conferencing Node. The status also shows the current media load on each Conferencing Node as a percentage of its total capacity.

When a Conferencing Node receives a call, it selects the node with the most available capacity in its location to handle the media for that conference. For more information, see Handling of media and signaling.

Backplane reservation

Each conference instance on each node reserves 1 HD connection for a backplane, to allow the conference to become geographically distributed if required. The exceptions to this are:

- Deployments with a single Conferencing Node. In such cases, no backplanes will ever be required, so capacity is not reserved.
- Conferences that are audio-only (in other words, where the conference has its **Conference capabilities** set to ***Audio-only***). In such cases, capacity equivalent to one audio connection is reserved for the backplane.
- Deployments with 1080p enabled. In such cases, backplanes reserve 1 Full HD connection of capacity, approximately double that of an HD connection.

For some reservation examples, see Resource allocation examples.

Gateway calls

Gateway calls (person-to-person calls) require sufficient capacity for both the inbound leg and the outbound leg. In general, this means that each gateway call consumes resources equivalent to two connections.

Non-distributed gateway calls involving only SIP or H.323 clients do not use any additional ports. However, in other scenarios, additional ports may be used:

- Distributed gateway calls (where the outbound leg is on a different Conferencing Node to the inbound leg) consume backplane ports — thus 1 HD video + 1 backplane for participant A plus 1 HD video + 1 backplane for participant B. This typically occurs when calling registered endpoints (where the outbound call to the registered endpoint will originate from the node the endpoint is registered to), or when using Call Routing Rules with an ***Outgoing location*** set to something other than ***Automatic***.
- For non-distributed gateway calls involving a Lync / Skype for Business client, a backplane is reserved in case the Lync/SfB user starts presenting (as the RDP/VbSS presentation stream could connect to any Conferencing Node due to DNS). Each presentation stream counts as 1 HD port. Thus if the incoming RDP/VbSS call lands on the same node as the video call then the resource usage is equivalent to 3 HD ports (2 video + 1 presentation). If the RDP/VbSS call lands on a different node to the video call then the resource usage for the call is equivalent to 5 HD ports (2 video + 2 backplane + 1 presentation).
- For calls between SIP/H.323 clients and an Infinity Connect WebRTC client, 1 port is reserved for a full motion presentation stream (so these calls will consume 3 HD ports). A call between 2 WebRTC clients will also reserve 3 HD ports (2 video + 1 full motion presentation) but could also use 1 extra port (making 4 in total) if the receiving client chooses to receive the presentation stream in full motion.

Licenses

The total number of concurrent calls that can be made in your deployment (regardless of whether those calls are HD, SD or audio-only) is limited by your license. For more information, see Pexip Infinity license installation and usage.

Implementing a dial plan

Before you deploy the Pexip Infinity platform in your network, you must consider your dial plan and ensure that calls to Virtual Meeting Room, Virtual Auditorium and Virtual Reception aliases, and that point-to-point calls handled by the Pexip Distributed Gateway, will be routed appropriately.

If your environment includes a PSTN gateway or uses an ITSP (Internet telephony service provider), consider the potential for toll fraud if you have Call Routing Rules that can route calls to the PSTN gateway or ITSP, or if you allow conference participants to dial out to other participants via the PSTN gateway or ITSP. See PSTN gateways and toll fraud for more information.

Service precedence

Incoming calls received by Pexip Infinity are routed as follows:

1. Pexip Infinity receives an incoming call via one of its Conferencing Nodes.
2. It checks whether the destination alias belongs to a Pexip Infinity Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service; if so, it directs the call to that service.
3. If the alias does not belong to any of the above services, Pexip Infinity checks the Call Routing Rules to see if the alias matches any rules specified there for incoming calls. If so, it places an outgoing call to the destination alias according to the rule's call target settings (which protocol and call control system to use, whether to route to registered devices only, etc).

This means that if an alias matches both a Virtual Meeting Room and a Call Routing Rule, the former will always take precedence and the call will be routed to the Virtual Meeting Room. You must therefore ensure that any regular expressions used in a Call Routing Rule do not unintentionally overlap with any aliases used by Virtual Meeting Rooms, Virtual Auditoriums, Virtual Receptions or Test Call Services.

Pexip Infinity's call routing logic is shown in the following diagram:

* This information can alternatively be provided by using external or local policy

For more information about incoming and outgoing Call Routing Rules and how to specify where calls are routed, see Configuring Call Routing Rules.

Using a standard format for aliases

We recommend that you use a standard format for all Virtual Meeting Room and Virtual Auditorium aliases within your Pexip Infinity deployment. A simple way to do this would be to have a set prefix such as **meet.**, followed by a user name, followed by your domain. For example:

- meet.alice.smith@example.com (for Alice's VMR)
- meet.bob.jones@example.com (for Bob's VMR)
- meet.sales@example.com (for the sales team's Virtual Auditorium)
- meet.reception@example.com (for the Virtual Reception)

Using this standard format will then make it simple to configure your call management system to route calls matching the format to the appropriate Conferencing Nodes.

Routing calls to the local Conferencing Node

When your call management system receives a call to a Virtual Meeting Room, Virtual Auditorium or Virtual Reception alias, it must then decide where to route the call.

A single conference can take place across one, two or more Conferencing Nodes with no difference in conference experience from the users' perspective. We recommend that in the first instance, calls are routed to the Conferencing Node that is geographically nearest to the endpoint that placed the call. This reduces the WAN bandwidth used for the conference in two ways: firstly, endpoints are only making local calls; and secondly, if more than one Conferencing Node is involved, only one set of media has to be sent between the two locations.

We recommend that you set up rules on your call management system so that calls originating from a particular location (defined by zone, subzone or IP address range) and placed to a Pexip Infinity alias are routed to a specific Conferencing Node. To build on the previous example, we have configured subzones on our call management system which group together endpoints in a particular physical location, and we have then set up the following call routing rules:

Source	Alias	Destination zone
Oslo subzone	meet.*@example.com.*	Norway Conferencing Node
New York subzone	meet.*@example.com.*	Americas Conferencing Node
Boston subzone	meet.*@example.com.*	Americas Conferencing Node

In this case, users in Oslo, New York and Boston could all call **meet.sales@example.com** and be routed to the same conference.

See Regular expression (regex) reference for information about writing regular expressions.

Further information

For further information about how to configure your specific call management system to work with Pexip Infinity, see the following documentation:

- **Pexip Infinity and Microsoft Lync / Skype for Business Deployment Guide**
- **Pexip Infinity and Cisco VCS Deployment Guide**
- **Pexip Infinity and Cisco Unified Communications Manager Deployment Guide**
- **Pexip Infinity and Polycom DMA Deployment Guide**

DNS record examples

This section describes the DNS records required for endpoints, federated Lync / Skype for Business environments and Infinity Connect clients to access Pexip Infinity services.

You can use the tool at http://dns.pexip.com to lookup and check SRV records for a domain.

Enabling endpoints to discover Conferencing Nodes

The following examples show the records that are required in DNS for endpoints to discover Pexip Infinity Conferencing Nodes.

It shows example DNS A and SRV records for the domain **example.com** to allow:

- SIP and H.323 endpoint registration messages to be routed to Pexip Infinity
- calls from non-registered SIP and H.323 endpoints to be routed to Pexip Infinity

If you have multiple Conferencing Nodes, you must set up DNS A and SRV records for each node.

Host DNS A-records

This example assumes there are 2 Conferencing Nodes, thus 2 DNS A-records are required.

Hostname	Host IP address
conf01.example.com.	198.51.100.40
conf02.example.com.	198.51.100.41

In your actual deployment, both the **Hostname** and **Host IP address** should be changed to use the real names and addresses of your Conferencing Nodes.

DNS SRV records

The following DNS SRV records are required for H.323 and SIP services. For each service there should be one record per host (Conferencing Node). Therefore, in this example, there are 2 records per service — one for each target host — **conf01.example.com** and **conf02.example.com**.

Name	Service	Protocol	Priority	Weight	Port	Target host
example.com. example.com.	h323cs h323cs	tcp tcp	10 10	10 10	1720 1720	conf01.example.com. conf02.example.com.
example.com. example.com.	h323ls h323ls	udp udp	10 10	10 10	1719 1719	conf01.example.com. conf02.example.com.
example.com. example.com.	h323rs h323rs	udp udp	10 10	10 10	1719 1719	conf01.example.com. conf02.example.com.
example.com. example.com.	sip sip	tcp tcp	10 10	10 10	5060 5060	conf01.example.com. conf02.example.com.
example.com. example.com.	sip sip	udp * udp *	10 10	10 10	5060 5060	conf01.example.com. conf02.example.com.
example.com. example.com.	sips sips	tcp tcp	10 10	10 10	5061 5061	conf01.example.com. conf02.example.com.

* SIP UDP is disabled on Pexip Infinity by default.

In your actual deployment, both the **Name** and the **Target host** should be changed to use the real domain name and host names of your Conferencing Nodes.

In these examples, the DNS records would be:

```
_h323cs._tcp.example.com. 86400 IN SRV 10 10 1720 conf01.example.com.
_h323cs._tcp.example.com. 86400 IN SRV 10 10 1720 conf02.example.com.

_h323ls._udp.example.com. 86400 IN SRV 10 10 1719 conf01.example.com.
_h323ls._udp.example.com. 86400 IN SRV 10 10 1719 conf02.example.com.

_h323rs._udp.example.com. 86400 IN SRV 10 10 1719 conf01.example.com.
```

```
_h323rs._udp.example.com. 86400 IN SRV 10 10 1719 conf02.example.com.

_sip._tcp.example.com.    86400 IN SRV 10 10 5060 conf01.example.com.
_sip._tcp.example.com.    86400 IN SRV 10 10 5060 conf02.example.com.

_sip._udp.example.com.    86400 IN SRV 10 10 5060 conf01.example.com.
_sip._udp.example.com.    86400 IN SRV 10 10 5060 conf02.example.com.

_sips._tcp.example.com.   86400 IN SRV 10 10 5061 conf01.example.com.
_sips._tcp.example.com.   86400 IN SRV 10 10 5061 conf02.example.com.

conf01.example.com.       86400 IN A 198.51.100.40
conf02.example.com.       86400 IN A 198.51.100.41
```

If the ability to implement geographically-aware DNS exists, then different weights and priorities could be set on the SRV records for different locations. For example, if there is both a European and a North American site, each with separate DNS servers, the European one could return European Pexip nodes at a higher priority than the North American nodes, and vice versa.

Enabling direct federation with remote Lync / Skype for Business environments

To ensure that calls from remote Lync / Skype for Business environments are routed to your Conferencing Node, a DNS SRV record in the format **_sipfederationtls._tcp.<domain>** must be created.

Example

Assume that the following **_sipfederationtls._tcp.example.com** DNS SRV and associated A-record have been created:

```
_sipfederationtls._tcp.example.com. 86400 IN SRV 1 100 5061 sip.example.com.
sip.example.com.                    86400 IN A 198.51.100.40
```

The SRV record points to the DNS A-record **sip.example.com** (this is a typical convention in Lync / Skype for Business environments), port 5061, with a priority of 1 and a weight of 100. In other words, it tells Lync / Skype for Business to send its TLS requests to **sip.example.com** on TCP port 5061. The A-record then resolves **sip.example.com** to the node's IP address (198.51.100.40).

In your actual deployment, the domain and host IP address should be changed to use the real name and address of your Conferencing Node.

Note that the domain name used in the SRV record has to match the domain in the corresponding A-record. This is required due to the trust model for Lync/SfB federation. Using our example:

- _sipfederationtls._tcp.**example.com** must use the same domain as sip.**example.com**
- you cannot, for example, configure the _sipfederationtls._tcp.example.com SRV record to point to sip.**video.example.com** or conf01.**otherdomain.com**).

For more information, see Pexip Infinity configuration for public DMZ deployments with remote Lync / Skype for Business environments.

Enabling access from the Infinity Connect Mobile client and the Infinity Connect desktop client

To enable participants to connect to conferences within your deployment using the Infinity Connect desktop client or Infinity Connect Mobile client, you must provide a DNS lookup so that these clients know which host to contact. The host will typically be a reverse proxy (for deployments where Conferencing Nodes are located within a private network), but it can also be a public-facing Conferencing Node.

To enable access from these desktop and mobile clients, each domain used in aliases in your deployment must either have an SRV record for **_pexapp._tcp.<domain>**, or resolve directly to the IP address of a reverse proxy or a public-facing Conferencing Node.

The SRV records for **_pexapp._tcp.<domain>** should always:

- point to an FQDN which **must** be valid for the TLS certificate on the target Conferencing Nodes or reverse proxy
- reference port 443 on the host.

Example

Assume that the following **_pexapp._tcp.example.com** DNS SRV records have been created:

```
_pexapp._tcp.example.com. 86400 IN SRV 10 100 443 proxy1.example.com.
_pexapp._tcp.example.com. 86400 IN SRV 20 100 443 proxy2.example.com.
```

These point to the DNS A-records **proxy1.example.com**, port 443 (HTTPS), with a priority of 10 and a weight of 100, and **proxy2.example.com**, port 443, with a relatively lower priority of 20 and a weight of 100.

- This tells the Infinity Connect desktop client to initially send its HTTP requests to host **proxy1.example.com** (our primary reverse proxy server) on TCP port 443. The desktop client will also try to use host **proxy2.example.com** (our fallback server) if it cannot contact proxy1.
- The Infinity Connect Mobile client will send its HTTP requests either to **proxy1.example.com** or to **proxy2.example.com**, depending on the order of the returned SRV records. If it fails to contact the first host, it will not attempt to contact the second host address.

For more information, see Setting up DNS records for Infinity Connect Mobile client and Infinity Connect desktop client use.

Next steps after initial installation

Making a test call

When you have deployed a Conferencing Node and configured a Virtual Meeting Room and an alias, you can make a test call to check that your system is working.

An easy way to do this is by using the Infinity Connect Web App to dial the alias of one of the Virtual Meeting Rooms you've already created, as follows:

1. Open a browser (we recommend Chrome) and type in the IP address (or FQDN, if you've set it up already) of one of the Conferencing Nodes.
2. In the **Person or conference to call** field, enter the alias of the VMR you want to use for testing.
3. Select **Connect** and check your microphone, camera and speakers are working as expected.
4. Select **Start**.
   You will be connected to the VMR.
5. From another machine, join the conference in the same way.

The two participants should be able to see and hear each other, and share content.

See About the Infinity Connect Web App for more information.

Further configuration

You are now ready to continue configuring the Pexip Infinity platform and services and deploying more Conferencing Nodes.

Specifically, you should now do the following:

- Assigning hostnames and FQDNs
- Enabling SNMP on Conferencing Nodes

At some point you may also want to:

- integrate the Pexip Infinity platform with your call control system
- configure the Pexip Distributed Gateway
- register devices directly to the Pexip Infinity platform
- customize the user experience

Integrating with a call control system

To integrate Pexip Infinity with your call control system, you must configure a trunk or neighbor zone towards each of the Conferencing Nodes.

For further information about how to configure your specific call management system to work with Pexip Infinity, see the following documentation:

- **Pexip Infinity and Microsoft Lync / Skype for Business Deployment Guide**
- **Pexip Infinity and Cisco VCS Deployment Guide**
- **Pexip Infinity and Cisco Unified Communications Manager Deployment Guide**
- **Pexip Infinity and Polycom DMA Deployment Guide**

Configuring the Pexip Distributed Gateway

The Pexip Distributed Gateway service enables endpoints to make calls to other devices or systems that use different protocols and media formats.

For more information, see About the Pexip Distributed Gateway service.

Registering devices directly to the Pexip Infinity platform

SIP and H.323 endpoints, and some Infinity Connect clients can register directly to Pexip Infinity Conferencing Nodes. This allows Pexip Infinity to route outbound calls to those registered devices without having to go via a SIP proxy or H.323 gatekeeper, or rely on DNS.

For more information, see Registering devices to Pexip Infinity.

Customizing the user experience

You can easily apply your own corporate branding to the Pexip Infinity platform, and produce a personalized user experience for all of your Pexip Infinity services.

For more information, see Customizing the Pexip Infinity user experience.

Administering Pexip Infinity

For information about how to administer Pexip Infinity, see:

Using the Pexip Infinity Administrator interface 73
Pexip Infinity system configuration 75
Pexip Infinity platform configuration 90
Conferencing Node configuration 118
Pexip Infinity services 142
Managing conferences 197
Customizing with themes 233
Pexip Infinity maintenance tasks 246
Pexip Infinity reference information 271

Using the Pexip Infinity Administrator interface

After you have run the installation wizard, all further configuration should be done using the Pexip Infinity Administrator interface.

This topic covers:

- Accessing the Pexip Infinity Administrator interface
- Setting the session timeout
- Changing the display language
- Timezones
- Getting help

Accessing the Pexip Infinity Administrator interface

The Pexip Infinity Administrator interface is hosted on the Management Node. To access this:

1. Open a web browser and type in the IP address or DNS name that you assigned to the Management Node using the installation wizard (you may need to wait a minute or so after installation is complete before you can access the Administrator interface).
2. Until you have uploaded appropriate TLS certificates to the Management Node, your browser may present you with a warning that the website's security certificate is not trusted. You should proceed, but upload appropriate TLS certificates to the Management Node (and Conferencing Nodes, when they have been created) as soon as possible.

   The **Pexip Infinity Conferencing Platform** login page will appear.
3. Log in using the web administration username and password you set using the installation wizard.

You are now ready to begin configuring the Pexip Infinity service and deploying Conferencing Nodes.

As a first step, we strongly recommend that you configure at least 2 additional NTP servers or NTP server pools to ensure that log entries from all nodes are properly synchronized. For details, see Enabling NTP on the Management Node.

For further information on the next steps, see Pexip Infinity system configuration.

Setting the session timeout

By default, Pexip Infinity Administrator interface sessions will time out after 30 minutes of inactivity. You can modify the **Management web interface session timeout** setting via the **Global Settings** page (**Platform Configuration > Global Settings**).

Changing the display language

The Pexip Infinity Administrator interface supports localization, meaning it can be displayed in different languages. The v14.1 release includes the following languages:

- English (default)
- Simplified Chinese
- Russian

Future releases will include other languages.

To select the language used for the Pexip Infinity Administrator interface, you must edit the language preferences within your browser settings. You must then refresh your browser for the changes to take effect.

You can edit your browser's language preferences as follows:

Chrome

1. Select the control button at the top right of the screen.
2. Select **Settings**.
3. Select **Show advanced settings**.
4. In the **Languages** section, select **Language and input settings...**.

Internet Explorer

1. Select the Tools button at the top right of the screen.
2. Select **Internet options**.
3. At the bottom of the window, select **Languages**.

Firefox

1. Select the Firefox button at the top left of the screen.
2. Select **Options > Options**.
3. Select the **Content** tab and in the **Languages** section, select **Choose...**.

Timezones

The Pexip Infinity Administrator interface displays all times (except for those in the Administrator log and Support log, and usage graphs) in your local time. This is obtained from the timezone configured on the computer you are using to access the Administrator interface. So for example, if you are sitting in an office in London and you log in to a Management Node that is located in New York, call start and end times will be shown in London time.

The timezone being used will always be shown in brackets after the time, in the format **2015-02-25 22:00:12 (GMT)**.

Logs (and usage graphs, which are based on the information in the logs) are always shown in UTC because of the distributed nature of the Pexip Infinity platform. Logs use the format **2015-02-24T23:16:33.019+00:00**.

Pexip Infinity Management Node and Conferencing Node VMs use the UTC timezone, and all logs are in UTC. Do not attempt to change the timezone on these systems.

Getting help

There are two types of help integrated into the Pexip Infinity Administrator interface:

Field-level help

By default, most configuration fields will show an explanation of the field underneath the input area. You can turn these explanations on or off by clicking on the ? icon on the bottom right of the page.

Context-sensitive help

From any page, clicking on the **Help** link at the top right of the page will open a new browser window showing the online **Pexip Infinity Administrator Guide**, which will be opened to the section that relates to the page you are on. From there you can use the table of contents on the left-hand side of the help window to navigate and view the entire guide. There is also a search box at the top right of the browser window which you can use to search for individual words or phrases.

More deployment information and PDF downloads of all documentation are available on the Pexip Infinity technical documentation website.

Contacting support

If you cannot find the information you require, contact your Pexip authorized support representative.

Pexip Infinity system configuration

Enabling DNS

At least one DNS server must be added to your system.

DNS is required when using hostnames instead of IP addresses when setting up NTP servers, syslog servers and VM managers to be used by Pexip Infinity. It is also used for call routing that depends on FQDNs.

To add, edit or delete the set of available DNS servers, go to **System Configuration > DNS Servers**.

The available options are:

Option	Description
Address	The IP address of the DNS server.
Description	An optional description of the DNS server.

After configuring the DNS servers available to your system, you should assign appropriate DNS servers to each location (**Platform Configuration > Locations**). Each Conferencing Node in that location will then use those DNS servers.

You can also select the specific DNS servers to be used by the Management Node (**Platform Configuration > Management Node**).

Enabling NTP on the Management Node

All host servers **must** be synchronized with accurate time before you install the Pexip Infinity Management Node or Conferencing Nodes on them.

You **must** enable NTP on your Management Node before deploying any Conferencing Nodes.

Pexip Infinity uses NTP servers to obtain accurate system time. This is necessary to ensure correct operation, including configuration replication and log timestamps.

Pexip Infinity Management Node and Conferencing Node VMs use the UTC timezone, and all logs are in UTC. Do not attempt to change the timezone on these systems. Note however that the administrator web interface uses your local time. For more information, see Timezones.

We strongly recommend that you configure at least 3 distinct NTP servers or NTP server pools to ensure that log entries from all nodes are properly synchronized.

To add, edit or delete the set of available NTP servers, go to **System Configuration > NTP Servers**.

The available options are:

Option	Description
Address	The IP address or FQDN of the NTP server.
Description	An optional description of the NTP server.
Secure NTP key ID	This optional field allows you to configure a key ID which is used in conjunction with the **Secure NTP key** for authenticating access to a secure NTP server.
Secure NTP key	This optional field, used in conjunction with the **Secure NTP key ID**, allows configuration of a SHA1 key for authenticating access to a secure NTP server. Enter the plaintext key; this will be stored as a SHA1 hash.

After configuring the NTP servers available to your system, you should assign appropriate NTP servers to each location (**Platform Configuration > Locations**). Each Conferencing Node in that location will then use those NTP servers.

You can also select the specific NTP servers to be used by the Management Node (**Platform Configuration > Management Node**).

Enabling SNMP

The Pexip Infinity Management Node and Conferencing Nodes can be monitored using SNMP. This topic covers:

- SNMP support in Pexip Infinity
- Adding SNMP NMSs
- Enabling SNMP on the Management Node
- Enabling SNMP on Conferencing Nodes

SNMP support in Pexip Infinity

You enable and disable SNMP support on each node individually.

You can optionally nominate an SNMP Network Management System (NMS) to receive trap notifications:

- If you enable SNMP on the Management Node, you can then nominate the NMS that will receive its notifications. For instructions see Enabling SNMP on the Management Node.
- If you enable SNMP on a Conferencing Node, you must configure the node's system location with an NMS. That NMS will receive notifications for all Conferencing Nodes in that location that have SNMP enabled. For instructions see Enabling SNMP on Conferencing Nodes.

Note that Pexip Infinity does not currently support traps with SNMPv3. If traps are required, use SNMPv2c.

Pexip Infinity supports SNMPv2c (non-secure) and SNMPv3 (secure) access to the basic RFC 1213 MIB-II tree (1.3.6.1.2.1) with read-only functionality. This includes full or partial support for:

- system_mib
- interfaces
- snmp_mib
- at
- ip
- icmp
- udp
- tcp
- RFC 1514/RFC 2790 MIB-II host MIB

When SNMP has been enabled on a Conferencing Node or Management Node, SNMP traps are sent to the selected SNMP NMS in a number of scenarios. The SNMP traps we support are:

Trap	OID	Description
cold start	1.3.6.1.6.3.1.1.5.1	Emitted when the snmpd service running on the node starts or restarts (due to snmp being reconfigured and/or due to the Conferencing Node rebooting).
authentication failure	1.3.6.1.6.3.1.1.5.5	Generated, for example, when any attempt to query SNMP values is made using an incorrect community string.
warm start	1.3.6.1.6.3.1.1.5.2	Generated when any software component fails unexpectedly (and coincides with the generation of a Pexip Incident Report).

The SNMP support in Pexip Infinity is built on top of the popular **net-snmp** open source implementation and therefore inherits some of the same behaviors (for example, generating a coldstart rather than warmstart on reconfiguration). For this reason you may also see some **net-snmp**-specific traps, such as the **nsNotifyShutdown** trap (OID 1.3.6.1.4.1.8072.4.0.2) when the snmpd daemon shuts down.

For some examples of how to use the Pexip Infinity Management API and SNMP to obtain information about the system (including how to obtain the CPU load average), see **Pexip Infinity Management API Guide**.

Adding SNMP NMSs

If you want SNMP notifications to be sent to a SNMP Network Management System (NMS) from a Management Node or Conferencing Node, you should add the details of the NMS to Pexip Infinity before you enable SNMP on the node.

To do this:

1. Go to **System Configuration > SNMP NMSs** and select **Add SNMP Network Management System**.
2. Complete the required fields:

Option	Description
Name	The name used to refer to this SNMP Network Management System
Description	An optional description of the SNMP Network Management System.
Address	The IP address or FQDN of the SNMP Network Management System.
Port	The SNMP port of the Network Management System. Default: 161.
SNMP trap community	The SNMP trap community name. Default: public

3. Select **Save**.

The NMS will now be available for selection on the **Edit Management Node** page when Enabling SNMP on the Management Node, and the **Edit System Location** page when Enabling SNMP on Conferencing Nodes.

Enabling SNMP on the Management Node

To enable SNMP on the Management Node:

1. Go to **Platform Configuration > Management Node** and select the Management Node.
   You will be taken to the **Edit Management Node** page.
2. Configure the SNMP settings for the Management Node. The options are:

Option	Description
SNMP mode	Configures the SNMP mode for the selected node: ***Off***: SNMP is disabled. You will not be able to use SNMP to query the node for its status. ***SNMPv2c read-only***: enables insecure, read-only access. ***SNMPv3 read-only***: enables secure, read-only access, using the **authPriv** security level with SHA1 authentication and AES 128-bit encryption. When enabled, access is provided to the basic RFC 1213 MIB-II tree (1.3.6.1.2.1). Default: ***Off***.
SNMP community	The SNMP group to which this node belongs. This setting applies to SNMPv2c only. Default: public
SNMPv3 username	The node's SNMPv3 username, used to authenticate SNMPv3 requests.
SNMPv3 privacy password	The node's SNMPv3 privacy password used for encrypting messages between the node and the management station. AES encryption must be used; DES is not supported.
SNMPv3 authentication password	The node's SNMPv3 authentication password, used to authenticate the associated username. The SHA authentication protocol must be used; MD5 is not supported.
SNMP system contact	The contact details (for example, email address) of the person responsible for this particular node.
SNMP system location	A description of the node's location.

3. If you want SNMP traps to be sent from the Management Node to an SNMP Network Management System (NMS), select the NMS from the **SNMP NMS** drop-down menu.
   If you have not already added the SNMP NMS, you can do so now by clicking ✚.
4. Select **Save**.

Enabling SNMP on Conferencing Nodes

To enable SNMP on a Conferencing Node:

1. Go to **Platform Configuration > Conferencing Nodes** and select the Conferencing Node.
   You will be taken to the **Edit Conferencing Node** page.
2. Configure the SNMP settings for the selected Conferencing Node. The options are:

Option	Description
SNMP mode	Configures the SNMP mode for the selected node: ***Off***: SNMP is disabled. You will not be able to use SNMP to query the node for its status. ***SNMPv2c read-only***: enables insecure, read-only access. ***SNMPv3 read-only***: enables secure, read-only access, using the **authPriv** security level with SHA1 authentication and AES 128-bit encryption. When enabled, access is provided to the basic RFC 1213 MIB-II tree (1.3.6.1.2.1). Default: ***Off***.
SNMP community	The SNMP group to which this node belongs. This setting applies to SNMPv2c only. Default: public
SNMPv3 username	The node's SNMPv3 username, used to authenticate SNMPv3 requests.
SNMPv3 privacy password	The node's SNMPv3 privacy password used for encrypting messages between the node and the management station. AES encryption must be used; DES is not supported.
SNMPv3 authentication password	The node's SNMPv3 authentication password, used to authenticate the associated username. The SHA authentication protocol must be used; MD5 is not supported.
SNMP system contact	The contact details (for example, email address) of the person responsible for this particular node.
SNMP system location	A description of the node's location.

3. Select **Save**.
4. If you enable SNMP on a Conferencing Node, you must ensure that the node's system location is configured with the NMS that will receive trap notifications from that node:
   a. Go to **Platform Configuration > Locations**.
   b. Select the Location to which the Conferencing Node belongs.
      You are taken to the **Edit System Location** page.
   c. From the **SNMP NMS** drop-down menu, select the Network Management System to which traps will be sent. Note that this NMS applies to all Conferencing Nodes in this location that have SNMP enabled.
      If you have not already added the SNMP NMS, you can do so now by clicking ✚.
   d. Select **Save**.

Using a syslog server

The Management Node acts as a syslog server, collating the logs from itself and each Conferencing Node to produce the Pexip Infinity support log. The support log includes the administrator log.

However, you can also use a remote syslog server to collect copies of each system's logs. The advantage of this is that, should the Management Node become unavailable, logging would still continue on the syslog server.

When a remote syslog server is used, the Management Node and each of the Conferencing Nodes sends its logs directly to the syslog server using the syslog protocol. The Management Node will still continue to collate logs from all the nodes, and it is these logs that will be shown from the Pexip Infinity Administrator interface.

To add, edit or delete the remote syslog servers used by Pexip Infinity, go to **System Configuration > Syslog Servers**.

The available options are:

Option	Description
Description	An optional description of the syslog server.
Address	The IP address or FQDN of the remote syslog server.
Port	The port on the remote syslog server. Default: 514.
Transport	The IP transport protocol used to communicate with the remote syslog server. Default: UDP.

Configuring SMTP servers

SMTP servers are used in conjunction with Sending provisioning emails to VMR and device owners when bulk-provisioning VMRs and device aliases from Active Directory via LDAP.

To add, edit or delete the SMTP servers used by Pexip Infinity, go to **System Configuration > SMTP Servers**.

The available options are:

Option	Description
Name	The name used to refer to the SMTP server.
Description	A description of the SMTP server.
Address	The IP address or FQDN of the SMTP server.
Port	The port on the SMTP server to connect to. Default: 587.
Username Password	These are optional fields where you can specify the credentials of a valid account on the SMTP server.
From email address	The "from" email address to use when sending emails via this server. This must be an email address that is permitted to be used for sending email using this server and account.
Connection security	The type of connection security to use when connecting to this email server. Select ***StartTLS*** to use an encrypted connection. Default: None.

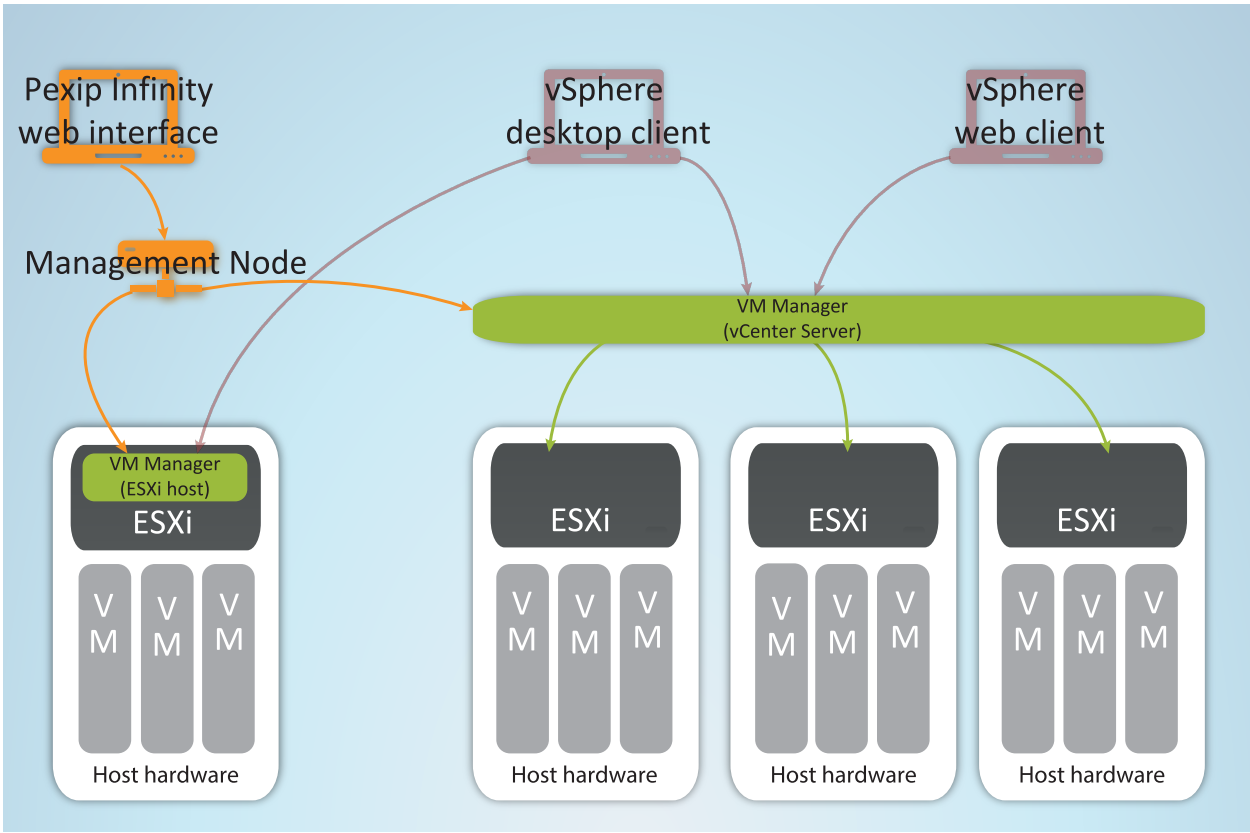
Using a VM manager

A VM manager is an application that allows you to connect to one or more VMware vSphere ESXi hypervisors. Pexip Infinity logs in to the selected VM manager when Automatically deploying a new Conferencing Node on a VMware host. It does not log in at any other time, and the credentials used to log in to the VM manager are not stored by Pexip Infinity.

Version 14.1 of the Pexip Infinity platform supports VMware vSphere ESXi 5.x and 6.0, although we recommend ESXi 5.5 or 6.0. Support for ESXi 4.1 is being deprecated - if you have upgraded from a version prior to v12, you can still deploy Conferencing Nodes to servers running ESXi 4.1; however if you have a new deployment using v12 or later and attempt to deploy a Conferencing Node to a server running ESXi 4.1, that node will go straight into maintenance mode. The VM managers used to connect to these hypervisors are either:

- **vCenter Server**, an application that allows you to use a single interface to manage one or more host servers running ESXi.
- **ESXi** directly on the host, used when managing a single host server running ESXi.

You can also log in to a VM manager directly yourself. To connect to vCenter Server, use either the vSphere desktop or web client; to connect directly to an ESXi host you can only use the vSphere desktop client.



Overview of Pexip Infinity, vSphere clients, VM Manager and ESXi host connectivity

When creating a Conferencing Node for automatic deployment on an ESXi host, you are asked to select the VM manager to be used. If the ESXi host on which you wish to deploy the Conferencing Node is managed by vCenter Server, you **must** select that vCenter Server as the VM Manager. Do not attempt to deploy the conferencing node directly onto the ESXi host (by selecting the ESXi host as the VM manager).

The **Add VM Manager** page allows you to create a list of VM managers in advance of creating your Conferencing Nodes. You can also add a VM manager during the process of creating a Conferencing Node.

Adding VM managers

To add, edit or delete VM managers, go to **System Configuration > VM Managers**.

The available options are:

Option	Description
Name	The name that you will use to refer to this VM manager.
Description	An optional field where you can provide more information about the VM manager.
Address	The IP address or FQDN of the VM manager. If you are using vCenter Server, this will be the IP address or FQDN of the vCenter Server. If you are not using vCenter Server, this will be the IP address or FQDN of ESXi on the host server.

If an ESXi host is being managed by vCenter Server, all administration must be performed via vCenter Server. Do not log in directly to the ESXi host; configuration changes made in this way may be lost. To ensure that ESXi hosts being managed by vCenter Server are accessible via vCenter Server only and are not directly accessible, you should put them in Lockdown mode. Lockdown mode forces all operations to be performed through vCenter Server.

Managing static routes

Static routes are additional configuration settings that permit routing of traffic to networks that are not accessible through the default gateway.

Typically, static routes are configured on Conferencing Nodes that are deployed in a DMZ, and where the default gateway on those nodes routes traffic out to the internet. The static routes would allow those nodes to communicate with Pexip Infinity nodes or other systems in the local, internal network. (See Network routing and addressing options for Conferencing Nodes for more information.)

- You must configure the static routes you want to use (via **System Configuration > Static Routes**) before you can apply them to a Conferencing Node or a Management Node.
- Static routes can be assigned to Conferencing Nodes during the automatic or manual deployment process, as well as after it has been deployed.
  - In some situations, for example where a Conferencing Node's default gateway is out to the public internet, a static route back to the Pexip Infinity platform must be applied to the Conferencing Node during its initial deployment phase (otherwise it will not be able to communicate with the Management Node and pick up its configuration).
- Static routes can only be assigned to a Management Node after it has been deployed.

The static routes take immediate effect after they have been assigned to a node. You do not have to restart the VM.

Configuring the set of available static routes

To configure the set of static routes that can be applied to a Conferencing Node or Management Node:

1. Go to **System Configuration > Static Routes**.
2. Select **Add Static route** and then configure the relevant destination and gateway addresses:

Option	Description
Name	A unique short name or description of the static route.
Destination network address	The IP address to be used in conjunction with the **Network prefix** to determine the network addresses to which this route applies.
Network prefix	The prefix length used in conjunction with the **Destination network address** to determine the network addresses to which this route applies. The prefix length can be in the range 0–32 for IPv4 addresses and 0–128 for IPv6 addresses. For example, use a **Destination network address** of **10.0.0.0** and a **Network prefix** of **8** to route addresses in the range **10.0.0.0** to **10.255.255.255**.
Gateway IP address	The IP address of the gateway to the network for this route.

3. Select **Save**.

Assigning a static route to an existing node

To assign a static route to an existing Conferencing Node or Management Node:

1. Ensure that the static route that you want to apply has already been configured (**System Configuration > Static Routes**).
2. Go to **Platform Configuration > Conferencing Node** or **Platform Configuration > Management Node** as appropriate:
3. Select the node to which you want to assign the static route.
4. In the **Static routes** section, select from the list of **Available Static routes** the routes to assign to the node, and then use the right arrow to move the selected routes into the **Chosen Static routes** list.
5. Select **Save**.

Assigning a static route to a new Conferencing Node

Before you start the deployment process as described in Deploying new Conferencing Nodes, ensure that the static routes that you want to apply have already been configured (**System Configuration > Static Routes**).

Then, while specifying the network configuration parameters to be applied to the new Conferencing Node, you will be able to include the static routes that you want to assign to that node.

Managing users and roles via LDAP

You can configure the Pexip Infinity platform to authenticate and authorize login accounts via a centrally managed LDAP-accessible server. This provides increased security and more flexibility than using the single administrator account that is installed by default.

Instead of authenticating the supplied username and password credentials against its own internal database, Pexip Infinity contacts the LDAP server to authenticate the user. It then uses the user's LDAP group attributes in combination with role mappings defined in Pexip Infinity to determine which Pexip Infinity features the user is authorized to access.

You can also configure the Pexip Infinity platform for client certificate authentication. This means that instead of logging in to the Pexip Infinity Administrator interface via the standard login page, users present (via their browser) a client certificate containing their user identification details. The validation of the presented certificate acts as the authentication phase and the username attributes in the certificate are used to determine which features the user is authorized to access.

The configuration described here applies to all user accounts connecting to the Pexip Infinity Administrator interface or the Pexip Infinity API. It does not apply to SSH connections. When using LDAP authentication, Pexip Infinity is configured by default to work with a Windows Active Directory LDAP server, but it can also be configured to work with other LDAP-accessible databases.

All usernames and passwords are case sensitive.

The following sections describe:

- Configuration summary for LDAP authentication
- Configuring how users are authenticated
- Configuring account roles
- Configuring LDAP roles
- Example: configuring permissions for an AD group
- Reinstating the local admin account

Configuration summary for LDAP authentication

To enable authentication and authorization via LDAP, you will need to configure both the LDAP database (if it is not already configured with user details) and the Pexip Infinity platform.

The LDAP database must be configured with:

- users' credentials
- groups that define the capabilities of the users.

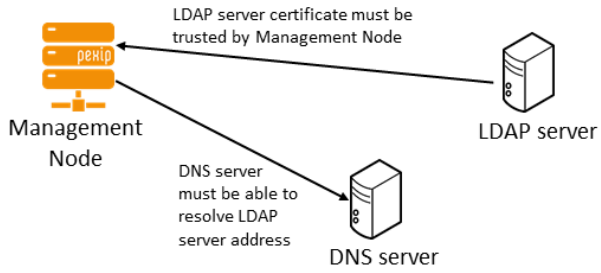
The Pexip Infinity platform must be configured with:

- an authentication source setting that uses an LDAP database
- connection details for the LDAP server; if the server address is an FQDN, ensure that it is resolvable over the DNS server configured for the Management Node
- account roles to control the actions that users can perform
- LDAP role mappings that map LDAP groups to account roles.

If a secure TLS connection between the LDAP server and the Management Node is required, ensure that:

- The LDAP server address is specified as an FQDN (so that it matches the name on the certificate presented by the LDAP server, which is typically created for the host name rather than the IP address).
- The Management Node trusts the certificate presented by the LDAP server; typically this means that the LDAP server certificate has to be uploaded to the Management Node as a trusted CA certificate (as the LDAP server's certificate is often generated by an internal authority which would not be included in Pexip's inbuilt list of trusted CA certificates).

  Note that the Management Node's server certificate does not have to be trusted by the LDAP server (unless the LDAP server has been explicitly configured to demand a client certificate).



The Pexip Infinity platform configuration steps for specifying an LDAP authentication source, and configuring account and LDAP role mappings are described in more detail in the following sections, and there is an example that shows how to configure permissions for an AD group. For information about installing server and trusted CA certificates, see Managing TLS and trusted CA certificates.

Configuring how users are authenticated

To configure how users are authenticated when they log in to the Pexip Infinity Administrator interface or API, go to **Users > User Authentication**. The options are:

Option	Description
Authentication source	The database to query for user authentication and authorization. ***Local database***: uses the Pexip Infinity local on-box database. Users can only log in via the default account (typically **admin**) and will have full administrator privileges. ***LDAP database***: users log in using an account configured on the LDAP database and obtain privileges according to the groups and roles associated with that account. Note that if this option is selected and the LDAP server is inaccessible for any reason, users will not be able to log in to the Pexip Infinity web-based Administrator interface or API. ***LDAP database and local database***: users can log in using either the default local **admin** account or via an account configured on the LDAP database. When using an LDAP database, you must configure the items in the **LDAP configuration** section. Default: ***Local database***.

Option	Description
Require client certificate	Controls whether all users log in to the Pexip Infinity Administrator interface via the standard login page or if they instead present (via their browser) a client certificate containing their user identification details. ***Not required***: users log in via the standard login page and provide a password which is authenticated against the selected **Authentication source**. ***Required (user identity in subject CN)***: users identify themselves via the identity contained in the subject CN (common name) of the client certificate presented by their browser. ***Required (user identity in subjectAltName userPrincipalName)***: users identify themselves via the identity contained in the **subjectAltName userPrincipalName** attribute of the client certificate presented by their browser. Default: ***Not required***. When a client certificate is required, the standard login page is no longer presented. Administrators will not be able to access the Pexip Infinity Administrator interface if their browser does not present a valid certificate that contains a user identity which exists in the selected **Authentication source**.
LDAP configuration	
LDAP server address	The domain name (for DNS SRV lookup), FQDN (for DNS A/AAAA lookup) or IP address of the LDAP server. If using a domain or an FQDN, ensure that it is resolvable over DNS. You must also ensure that Pexip Infinity has trusted CA certificates for the authority that signed the LDAP server's certificate (if a TLS connection is required). We strongly recommend that you do not use an IP address. If an IP address is used, and a TLS connection is required, this will only work if the IP address is specified as the common name in the LDAP server's certificate. See Troubleshooting LDAP server connections for more information about how the system establishes a connection to the LDAP server and how to troubleshoot connection issues.
Allow insecure transport	By default the system will attempt to establish a secure TLS connection with the LDAP server. Select this option if you want to allow the system to fall back to a TCP connection (using SASL DIGEST-MD5). You cannot specify the LDAP server by IP address if this option is selected.
LDAP bind username and password	The username and password of the bind account on the LDAP server. This should be a domain user service account, not the Administrator account.
LDAP base DN	The base DN (distinguished name) of the LDAP forest to query (e.g. **dc=example,dc=com**).
Advanced LDAP configuration By default the advanced LDAP configuration settings are preconfigured for Windows Active Directory, and may also be appropriate for other LDAP databases such as OpenLDAP.	
Search global catalog	Select this option to expand the scope of the search to the entire Active Directory Global Catalog instead of traditional LDAP.
LDAP user search DN	The DN relative to the **LDAP base DN** to query for user records (e.g. **ou=people**). If blank, the **LDAP base DN** is used. In deployments with large user bases, you may want to configure this to optimize the LDAP user queries.
LDAP user filter	The LDAP filter used to match user records in the directory. Default: (&(objectclass=person)(!(objectclass=computer)))
LDAP user search filter	The LDAP filter used to find user records when given the user name. The filter may contain {username} to indicate locations into which the username is substituted. This filter is applied in conjunction with the **LDAP user filter** and must contain at least one substitution. If client certificate-based authentication is used, this filter usually must include 'userPrincipalName={username})' either in addition to, or instead of, the default value; for example '(|(uid={username})(sAMAccountName={username})(userPrincipalName={username}))'. Default: (|(uid={username})(sAMAccountName={username}))

Option	Description
LDAP group attributes	A comma-separated list of attributes in the LDAP user record to examine for group membership information. The attribute value must contain the DN of each group the user is a member of. If no attributes are specified, or none of the specified attributes are present in the LDAP user record, an LDAP group search (using the remaining advanced configuration options below) will be performed instead. Default: memberOf
LDAP group search DN	The DN relative to the **LDAP base DN** to query for group records (e.g. **ou=groups**) when no group attributes are present in the LDAP user record. If blank, the **LDAP base DN** is used. In deployments with large user bases, you may want to configure this to optimize the LDAP group queries.
LDAP group filter	The LDAP filter used to match group records in the directory. Default: (|(objectclass=group)(objectclass=groupOfNames)(objectclass=groupOfUniqueNames)(objectclass=posixGroup))
LDAP group membership filter	The LDAP filter used to search for group membership of a user. The filter may contain **{userdn}** to indicate locations into which the user DN is substituted. The filter may contain **{useruid}** to indicate locations into which the user UID is substituted. This filter is applied in conjunction with the LDAP group filter and must contain at least one substitution. Default: (|(member={userdn})(uniquemember={userdn})(memberuid={useruid}))

If authentication against an LDAP database is configured, you can save the settings only if Pexip Infinity can successfully contact the specified LDAP server.

Note that all LDAP distinguished names must be entered as per the LDAP standard (RFC 4514). LDAP configuration is case insensitive.

Supporting nested security groups in Windows Active Directory

The default LDAP configuration does not support nested security groups in Windows Active Directory. For example, if group A is allowed to log in via LDAP, and if group B is a member of group A, then any user who is only a member of group B will not be allowed to log in.

To allow members of a nested Active Directory security group to log in over LDAP:

1. Go to **Users > User Authentication** and expand the **Advanced LDAP configuration** section.
2. Ensure that **LDAP group attributes** is empty (i.e. remove the default "memberOf" content).
3. Change **LDAP group membership filter** to "(member:1.2.840.113556.1.4.1941:={userdn})"
4. Select **Save**.

(This configuration uses the LDAP_MATCHING_RULE_IN_CHAIN OID. More information on this can be found at https://msdn.microsoft.com/en-us/library/aa746475%28VS.85%29.aspx.)

Configuring account roles

Account roles are used to control the actions that users can perform in the web-based Administrator interface or API after they have been authenticated. For example, you can configure a role that allows a user to only view (and not modify) specific items of configuration data or status information via the Administrator interface.

When a user has restricted permissions, all navigation menu options are still displayed to that user, but they are given an **Access denied** message if they try to select a menu option that they are not authorized to use. For read-only restrictions, the relevant **Add <item>** options are not displayed.

Two roles are present by default:

- ***Read-only***: allows read-only access to all configuration settings and status information when accessing the system via the web-based Administrator interface or the API.
- ***Read-write***: allows full administrative access when accessing the system via the web-based Administrator interface or the API.

To add, edit or delete account roles, go to **Users > Account Roles**. When configuring account roles, the options are:

Option	Description
Name	A descriptive name of the role, e.g. "auditor" or "management system".
Permissions	Select from the list of **Available permissions** the set of permitted actions for the role and then use the right arrow to move the selected actions into the **Chosen permissions** list. All roles must include the ***Is an administrator*** permission for access to the system. In addition, the ***May use web interface*** and ***May use API*** permissions must be included for access via the web-based Administrator interface and API respectively. You must then also add all of the other permissions, such as ***May modify system configuration*** and so on, that you want to apply to the role — if a role has, for example, only the ***Is an administrator*** and ***May use web interface*** permissions, a user with that role will be able to log in via the web-based Administrator interface but will not be able to perform any actions.

Configuring LDAP roles

LDAP roles are used to map the LDAP groups associated with LDAP user records to the Pexip Infinity account roles. You must configure a separate LDAP role for each LDAP group for which you want to map one or more Pexip Infinity account roles.

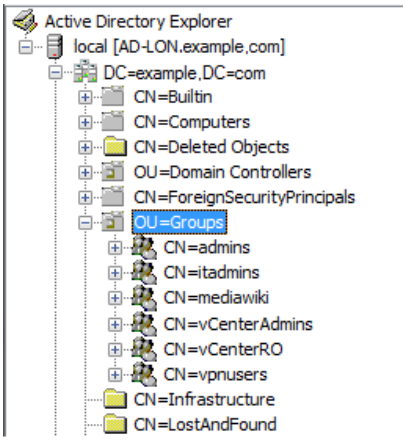
To add, edit or delete LDAP roles, go to **Users > LDAP Roles**. When configuring LDAP roles, the options are:

Option	Description
Name	A descriptive name of the role, e.g. "domain administrator with full privileges".
LDAP group DN	Select the LDAP group against which you want to map one or more account roles. The list of LDAP groups is only populated when there is an active connection to an LDAP server (**Users > User Authentication**).
Roles	Select from the list of **Available roles** the account roles to associate with the LDAP group and then use the right arrow to move the selected roles into the **Chosen Roles** list. All of the underlying permissions within a role are "positive" permissions, i.e. they allow the user to do something. If more than one role is selected, all of the permissions associated with each role are combined and granted to the relevant user. Note that you can select ✚ which will open a new window from where you can configure a new account role. When you save the role it is automatically added to the set of **Chosen Roles**.

Example: configuring permissions for an AD group

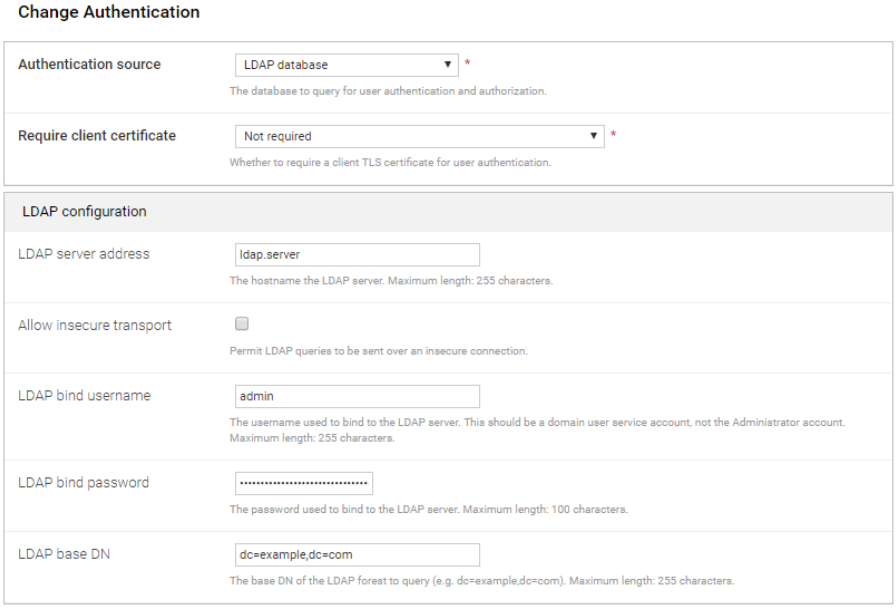
This example shows how you can configure the specific actions (permissions) that all members of an AD group are allowed to perform when administering Pexip Infinity.

Let's assume that you have the following set of groups already configured in Windows Active Directory:



This example shows how to configure all AD users who are members of the "itadmins" group to be able to add, modify and delete VMR/conference related settings, but only be able to view other configuration aspects of Pexip Infinity (system settings, logs etc).

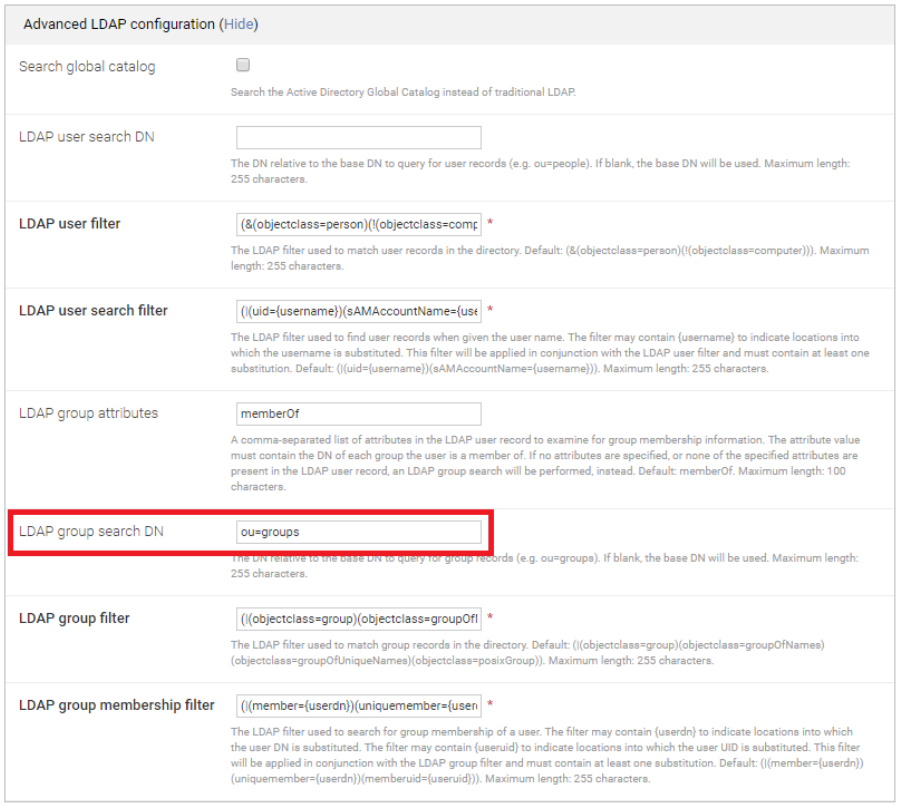
1. Ensure that you have configured an LDAP authentication source (**Users > User Authentication**) that can access your AD server, for example:



2. In this case, we want to define permissions based upon membership of specific AD groups, therefore we have configured the **LDAP group search DN** in the **Advanced LDAP configuration** settings to **ou=groups**.

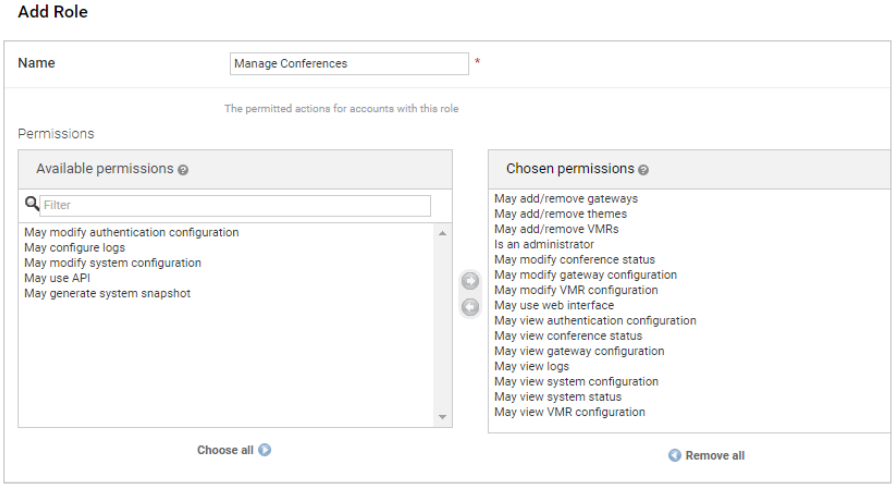
   This means that when we configure the LDAP roles, the set of LDAP groups that is presented will be filtered to include only those in the **groups** organizational unit (ou).

   This filtering step is not mandatory but it does make it easier to select the appropriate LDAP groups, and can optimize system performance.



3. We now need to configure an account role (**Users > Account Roles**) that defines the set of actions that can be performed by users who have been assigned that role.

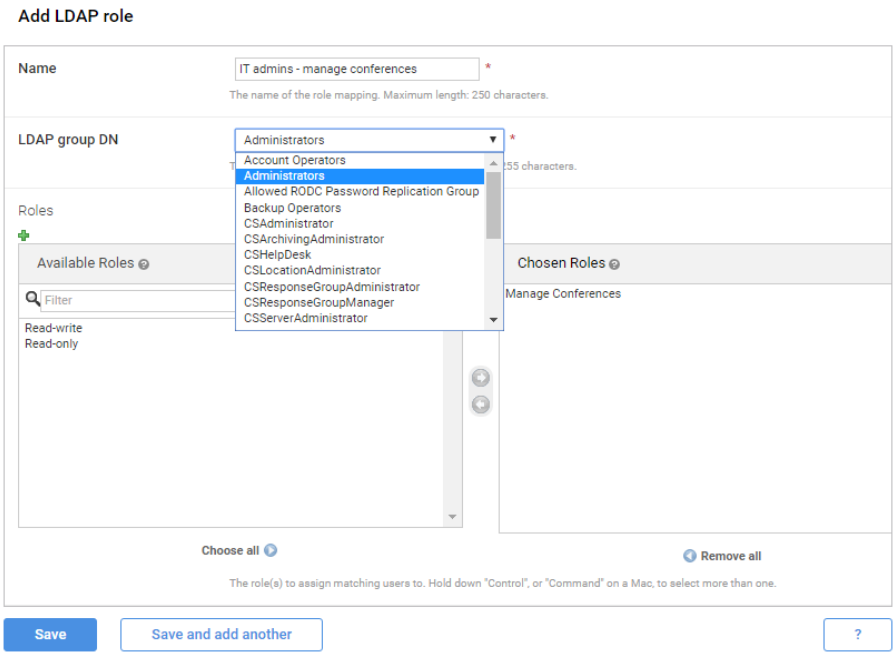
   Here, a "Manage Conferences" role has been created. The **Chosen permissions** allow a user to use the web interface to configure all service-related items such as VMRs, themes, gateway rules, but to only be able to view (and not modify) all other configuration.



4. The final step is to associate this account role with an LDAP role/group (**Users > LDAP Roles**).

   Here, we have configured an "IT admins - manage conferences" role. The **LDAP group DN** dropdown presents a list of LDAP groups from AD. In our case this list is filtered to only show those groups in the **ou=groups** organizational unit (due to the **LDAP group search DN** configuration in step 2).

   We have selected the **itadmins** group and associated it with the **Manage Conferences** role we created in step 3. (Note that you can associate the LDAP role with more than one account role if required.)



This means that AD users who are in the itadmins group will now be able to sign in to the Pexip Infinity Administrator interface, using their AD credentials, and configure service-related settings only.

To set up different permissions for members of other AD groups, repeat steps 3 and 4 to create different account role and LDAP role associations.

Reinstating the local admin account

If necessary you can reinstate access via the Pexip Infinity local on-box database, so that users can log in via the default account (typically **admin**) and will have full administrator privileges. You may need to do this if, for example the **Authentication source** is configured as ***LDAP database*** and your connectivity to the LDAP server goes down or your credentials become invalid.

To reactivate your local admin account:

1. Log in to the Management Node over SSH.
2. For local admin access only, run the command:
   `authset LDAP LOCAL`
   or, for LDAP and local admin access, run the command:
   `authset LDAP BOTH`

You can also disable client certificate authentication so that you can log in to the Pexip Infinity Administrator interface via the standard login page.

To disable certificate-based authentication:

1. Log in to the Management Node over SSH.
2. Run the command:
   `authset CBA OFF`

If you forget the password for the Pexip Infinity Administrator interface, you can re-run the installation wizard, being sure to change only the **Web administration password** setting.

Pexip Infinity platform configuration

Configuring the Management Node

The Management Node is a virtual machine which controls the Pexip Infinity service. It is used to deploy and manage one or more Conferencing Nodes, and configure the Pexip Infinity services (Virtual Meeting Rooms, Virtual Receptions and so on) that exist on those Conferencing Nodes. It also collates status and logging information about conferences across all Conferencing Nodes.

Initial configuration of the Management Node was provided at the time the software image was initially installed using the installation wizard, but if necessary you can edit some of this information afterwards.

All changes to the configuration of the Management Node should be done using the Pexip Infinity Administrator interface. Do not make any changes by any other means such as VMware or SSH; doing so may cause the Pexip Infinity service to fail. In particular you must not change the IP address of the Management Node.

To change configuration of the Management Node, go to **Platform Configuration > Management Node**.

The available options are:

Option	Description
Name	The name used to refer to this Management Node. It is made up of the DNS **Hostname** and **Domain suffix** that were assigned to the Management Node during initial installation (when the installation wizard was run).
Description	An optional field where you can provide more information about the Management Node. This will default to the **Name** but you can change it here.
DNS servers	From the list of configured DNS servers, select one or more DNS servers to be used by this Management Node.
NTP servers	From the list of configured NTP servers, select one or more NTP servers to be used by this Management Node.
IPv6 address	The IPv6 address for this Management Node.
Gateway IPv6 address	The IPv6 address of the default gateway.
Static routes	From the list of **Available Static routes**, select the routes to assign to the node, and then use the right arrow to move the selected routes into the **Chosen Static routes** list. For more information, see Managing static routes.

You can also change the Management Node's SNMP settings (see Enabling SNMP for more information):

Option	Description
SNMP mode	Configures the SNMP mode for the selected node: ***Off***: SNMP is disabled. You will not be able to use SNMP to query the node for its status. ***SNMPv2c read-only***: enables insecure, read-only access. ***SNMPv3 read-only***: enables secure, read-only access, using the **authPriv** security level with SHA1 authentication and AES 128-bit encryption. When enabled, access is provided to the basic RFC 1213 MIB-II tree (1.3.6.1.2.1). Default: ***Off***.
SNMP community	The SNMP group to which this node belongs. This setting applies to SNMPv2c only. Default: public
SNMPv3 username	The node's SNMPv3 username, used to authenticate SNMPv3 requests.
SNMPv3 privacy password	The node's SNMPv3 privacy password used for encrypting messages between the node and the management station. AES encryption must be used; DES is not supported.

Option	Description
SNMPv3 authentication password	The node's SNMPv3 authentication password, used to authenticate the associated username. The SHA authentication protocol must be used; MD5 is not supported.
SNMP system contact	The contact details (for example, email address) of the person responsible for this particular node.
SNMP system location	A description of the node's location.

If you want SNMP traps to be sent from the Management Node to a particular SNMP Network Management System (NMS), select the NMS from the **SNMP NMS** drop-down menu:

Option	Description
SNMP NMS	The Network Management System to which SNMP traps will be sent from the Management Node. If you have not already added the SNMP NMS, you can do so now by clicking ✚. Note that Pexip Infinity does not currently support traps with SNMPv3. If traps are required, use SNMPv2c.

Other details of the Management Node that cannot be changed via the Administrator interface are also shown on this page for your information, as follows:

Option	Description
IP address	The IPv4 address of the Management Node.
Network mask	The IPv4 network mask of the Management Node.
Gateway IP address	The IPv4 address of the default gateway.
Hostname	The DNS hostname of this Management Node.
Domain	The DNS domain of this Management Node.

If you need to change any of the hostname/addressing information, you must do so by Re-running the installation wizard.

To perform other maintenance tasks such as changing the IP address of the Management Node or moving the Management Node to a different host server, see Moving, restoring or changing the IP address of the Management Node.

About global settings

Platform-wide configuration changes are made using the **Global Settings** page (**Platform Configuration > Global Settings**).

The settings are:

Setting	Description	More information
Service configuration		
Guests-only timeout	The length of time (in seconds) for which a conference will continue with only Guest participants, after all Host participants have left. Default: 60 seconds	Using PINs to differentiate between Hosts and Guests
Last participant backstop timeout	The length of time (in seconds) for which a conference will continue with only one participant remaining. The type of participant (Host, Guest, automatically dialed, streaming etc) is irrelevant. The time can be configured to values between 60 seconds and 86400 (1 day), or to 0 (never eject). Default: 0 (never eject)	Automatically ending a conference

Setting	Description	More information
Default theme	The theme to use for services that have no specific theme selected.	Customizing images and voice prompts using themes
Maximum inbound call bandwidth (kbps)	Allows you to limit the bandwidth of media being received by Pexip Infinity from individual participants, for calls where bandwidth limits have not otherwise been specified.	Restricting call bandwidth
Maximum outbound call bandwidth (kbps)	Allows you to limit the bandwidth of media being sent by Pexip Infinity to individual participants, for calls where bandwidth limits have not otherwise been specified.	Restricting call bandwidth
Connectivity		
Enable SIP *	This allows you to enable or disable support for the SIP protocol over TCP and TLS across all Conferencing Nodes in your Pexip Infinity deployment. Note that disabling SIP will disable support for Lync / Skype for Business (MS-SIP). Default: enabled.	Enabling and disabling SIP, H.323, WebRTC and RTMP
Enable SIP UDP *	This allows or prevents incoming calls over SIP UDP. Default: disabled.	
Enable H.323 * Enable WebRTC * Enable RTMP *	These boxes allow you to enable or disable support for the selected protocol across all Conferencing Nodes in your Pexip Infinity deployment. Default: all of these settings are enabled by default.	
Enable 1080p (Full HD)	Enables the use of 1080p (Full HD) in calls. Default: disabled.	Enabling Full HD (1080p)
Enable chat	Enables relay of chat messages between conference participants using Lync / Skype for Business and Infinity Connect clients. Default: enabled.	Enabling and disabling chat
Enable outbound calls	Enables calls via the Pexip Distributed Gateway, and allows dial-out from Pexip Infinity using Pexip Infinity Connect clients, the Infinity Connect Mobile clients for iOS and Android, and the Pexip Infinity Administrator interface. Default: enabled.	About the Pexip Distributed Gateway service Manually dialing out to a participant from a conference
Enable support for Pexip Infinity Connect and Mobile App	Enable or disable support for Pexip Infinity clients. These include the Infinity Connect Web App, Infinity Connect desktop client, the Infinity Connect Mobile client for iOS or Android, and any other third-party applications that use the client API. Default: enabled.	Introduction to Infinity Connect Enabling and disabling Infinity Connect clients
Pexip Infinity domain (for Lync / Skype for Business integration)	The name of the SIP domain that is routed from Lync / Skype for Business to Pexip Infinity, either as a static route or via federation. You can also configure the Pexip Infinity domain on a per-location basis, which would override this global setting for Conferencing Nodes in that location.	**Pexip Infinity and Microsoft Lync / Skype for Business Deployment Guide**
Enable Lync / Skype for Business auto-escalation	When selected, this automatically escalates a Lync / Skype for Business audio call so that it receives video from a conference. Default: disabled.	Automatically escalating Lync / Skype for Business audio calls

Setting	Description	More information
Enable Lync / Skype for Business Video-based Screen Sharing (VbSS)	Enables and disables support for Skype for Business Video-based Screen Sharing (VbSS). This is a "tech preview" feature and is currently only supported when the Skype for Business client is either calling another endpoint via the Pexip Distributed Gateway, or calling into a Virtual Meeting Room or Virtual Auditorium. Default: disabled.	For information about enabling VbSS on your Skype for Business infrastructure see https://technet.microsoft.com/en-us/library/mt756736.aspx.
DSCP value for management traffic	The DSCP value for SSH, HTTPS and SNMP management traffic sent from the Management Node and from Conferencing Nodes. This is an optional setting used to prioritize different types of traffic in large, complex networks. Also see Configuring system locations.	
Enable SSH	Allows an administrator to log in to the Management Node and Conferencing Nodes over SSH. Default: enabled.	
Enable directory	When disabled, Infinity Connect clients will display aliases from their own call history only. When enabled, registered Infinity Connect clients will additionally display the aliases of Virtual Meeting Rooms, Virtual Auditoriums, Virtual Receptions, and devices registered to the Pexip Infinity deployment. Default: enabled.	Directory (phone book) of devices and VMRs for registered Infinity Connect clients
Port ranges		
Signaling port range start and end *	The start and end values for the range of ports (UDP and TCP) that all Conferencing Nodes use to send signaling (for H.323, H.245 and SIP). Default: 33000–39999.	
Media port range start and end *	The start and end values for the range of ports (UDP and TCP) that all Conferencing Nodes use to send media for H.323, SIP, Lync / Skype for Business, WebRTC and RTMP (note that RTMP uses TCP only). Default: 40000–49999.	
Codecs		
Codecs	Choose which codecs to offer in audio/video negotiation (SDPs). Some third-party systems can experience issues if they are sent a large SDP from Pexip Infinity. You can reduce the size of the SDP by disabling specific, unwanted codecs. Default: all codecs are selected except ***AAC-LD128***, ***H.264 High (mode 0)*** and ***H.264 High (mode 1)***. To enable the H.264 High Profile codec, move ***H.264 High (mode 1)*** into the list of **Chosen Codecs**. For optimal interoperability results, only enable ***H.264 High (mode 1)*** — leave ***H.264 High (mode 0)*** in the **Available Codecs** list.	

Setting	Description	More information
Security		
OCSP state	Whether to use OCSP when checking the validity of TLS certificates. ***Off***: OCSP checking is disabled. ***On***: An OCSP request will be sent to the URL specified in the TLS certificate. ***Override***: An OCSP request will be sent to the URL specified in the **OCSP responder URL** field. Default: ***Off***.	Using OCSP to check the status of certificates
OCSP responder URL	The URL to which OCSP requests will be sent either: • if the **OCSP state** is set to ***Override***, or • if the **OCSP state** is set to ***On*** but there is no URL specified in the TLS certificate.	Using OCSP to check the status of certificates
SIP TLS certificate verification mode	Determines whether to verify the peer certificate for connections over SIP TLS. ***Off***: the peer certificate will be not be verified; all connections will be allowed. ***On***: the peer certificate will be verified, and the peer's remote identities (according to RFC5922) will be compared against the Application Unique String (AUS) identified by Pexip Infinity before the connection is allowed. Default: ***Off***.	Verifying peer certificates for SIP TLS connections
External system integration		
Enable HTTP access for external systems	Access for external systems is over HTTPS by default. If this box is selected, access over HTTP is also permitted. Default: disabled.	Integrating with external systems
External system username and password	The username and password used by external systems (such as CUCM) when authenticating with Pexip Infinity.	Integrating with external systems
Administrator interface configuration		
Login banner text	Any text entered here is displayed in a message box on the login page. This field supports plain text only.	
Management web interface session timeout	The number of minutes a browser session may remain idle before the user is logged out of the Management Node Administrator interface. Default: 30 minutes.	
Reporting		
Enable incident reporting Incident reporting URL	If incident reporting is enabled, reports will be sent to the specified URL. This setting is enabled or disabled during initial installation of the Management Node (when running the installation wizard).	Automatically reporting errors
Automatically send deployment and usage statistics to Pexip	Select this option to allow submission of deployment and usage statistics to Pexip. This will help us improve the product. This setting is enabled or disabled during initial installation of the Management Node (when running the installation wizard).	Automatically sending usage statistics

Setting	Description	More information
Automatic backups		
Enable automatic backups	Configures Pexip Infinity to automatically backup the Management Node configuration data on a daily basis.	Backing up and restoring configuration
Backup passphrase	The text entered here is used to encrypt the backup file. You must remember this text as it will be required if you need to subsequently restore the data from the file.	
Cloud bursting		
Enable bursting to the cloud AWS access key ID / secret key Bursting threshold	These options enable and configure the Pexip Infinity platform for dynamic cloud bursting. The bursting threshold controls when your overflow nodes in AWS are automatically started up so that they can provide additional conferencing capacity. When the number of additional HD calls that can still be hosted in a location reaches or drops below the threshold, it triggers Pexip Infinity into starting up an overflow node in the overflow location.	Dynamic bursting to the AWS cloud

* If you change any of these values, each Conferencing Node must be rebooted before the changes will take effect.

About system locations

System locations are labels that allow you to group Conferencing Nodes together, typically according to where they are physically located. System locations serve two purposes:

- They enable Pexip Infinity to make intelligent decisions about routing of media and signaling, including overflowing to another location when a particular location reaches capacity.
- External services such as DNS, NTP, H.323 gatekeepers, SIP proxies, Lync / Skype for Business Servers, TURN servers, STUN servers, SNMP NMSs and policy profiles are configured on a per-location basis, and are used by all Conferencing Nodes in that location.

The Conferencing Node's system location is assigned when the node is deployed, however you can subsequently change its location.

If you change the system location of an existing Conferencing Node, all existing calls will be disconnected and the Conferencing Node will be restarted.

Intelligent routing

Grouping Conferencing Nodes by location allows Pexip Infinity to make intelligent decisions about routing. For example, if a single conference is taking place across groups of Conferencing Nodes in two different locations, then Pexip Infinity will nominate one Conferencing Node in each location to act as the proxy for that location. Media streams will be sent between each proxy only, rather than multiple streams being sent between each of the Conferencing Nodes at each of the locations. For more information, see Handling of media and signaling.

If a location reaches its capacity for a conference instance, additional "overflow" locations can be utilized to handle the media for new conference participants that are connected (for signaling purposes) to nodes within that location. For more information, see Location capacity overflow.

When grouping Conferencing Nodes into different locations, you should consider the amount of packet loss within your network. If there is a chance of packet loss due to network congestion between different groups of Conferencing Nodes by design, then they should be assigned separate system locations even if they are in the same physical location.

Network deployment considerations

Nodes within a location actively synchronize with each other and may require a relatively high amount of network bandwidth for their communication. Pexip's backplane topology for distributed conferences assumes that there is a high availability, low latency, high-bandwidth network link between nodes within a location.

Therefore, while physical proximity is not a requirement, nodes in the same system location should typically be in the same physical datacenter or in physically proximate datacenters with an excellent link between them.

If (as is often the case) you do not have such a network between your datacenters, we recommend that you consider assigning your nodes to different system locations. There should be no more than 150 ms latency between Conferencing Nodes.

In addition:

- Conferencing Nodes in a DMZ must not be configured with the same **System location** as nodes in a local network. This is to ensure that load balancing is not performed across nodes in the DMZ and nodes in the local network.
- Any Conferencing Nodes that are configured with a static NAT address must not be configured with the same **System location** as nodes that do not have static NAT enabled. This is to ensure that load balancing is not performed across nodes servicing external clients and nodes that can only service private IP addresses.

See Network deployment options for more information about the various deployment scenarios.

Configuring external services

DNS and NTP servers

When setting up a system location you must select at least one DNS server and at least one NTP server to be used by all of the Conferencing Nodes in that location. This allows you to, for example, assign nodes in a DMZ to a location that uses different DNS and NTP servers to a location containing nodes in a local, internal network.

H.323 gatekeepers, SIP proxies and Lync / Skype for Business servers

When setting up a system location you can optionally specify the H.323 gatekeeper, the SIP proxy, and the Lync / Skype for Business server / MSSIP domain to be used for outbound calls from that location. These are the call control systems that will be used when:

- a conference participant uses an Infinity Connect client to add another participant to the call
- the administrator manually dials out to a participant from a Virtual Meeting Room
- a participant is automatically dialed out to from a Virtual Meeting Room
- a third party uses the API to place a call to a participant

For more information, see About H.323 gatekeepers and SIP proxies and About Lync / Skype for Business servers.

TURN servers and STUN servers

Pexip Conferencing Nodes can utilize a TURN server and negotiate TURN relays with the following ICE capable clients:

- Lync / Skype for Business clients
- Infinity Connect WebRTC clients (Web App on Chrome, Firefox and Opera, and the mobile and desktop clients)

If these endpoints will be connecting to privately-addressed "on-premises" Conferencing Nodes, you **must** configure Pexip Infinity with the address of at least one TURN server that it can offer to ICE clients.

In some deployment scenarios where the TURN server is not located outside of the enterprise firewall, you may need to configure a separate STUN server so that each Conferencing Node can discover its public NAT address.

You can also configure the STUN servers to be used by Infinity Connect WebRTC clients when they connect to a Conferencing Node in this location.

For more information, see Using TURN servers with Pexip Infinity and Using STUN servers with Pexip Infinity.

SNMP NMSs

If you have enabled SNMP support on one or more Conferencing Nodes in a particular location, you must also select the SNMP Network Management System (NMS) to which SNMP traps will be sent. The selected NMS will be used for all Conferencing Nodes in this location that have SNMP support enabled.

Note that Pexip Infinity does not currently support traps with SNMPv3. If traps are required, use SNMPv2c.

For more information, see Enabling SNMP.

Policy profiles

Policy profiles are used to configure how Pexip Infinity uses external policy and/or local policy to control Pexip Infinity's call policy and routing decisions.

You can configure Pexip Infinity to use a different policy profile per system location.

For more information, see **Using external and local policy to control Pexip Infinity behavior**.

Configuring system locations

To add, edit or delete system locations, go to **Platform Configuration > Locations**.

The available options are:

Option	Description
Name	The name you want to give to this physical location.
Description	An optional field where you can provide more information about the location.
DNS servers	From the list of configured DNS servers, select one or more DNS servers to be used by all the Conferencing Nodes in this location.
NTP servers	From the list of configured NTP servers, select one or more NTP servers to be used by all the Conferencing Nodes in this location.
H.323 gatekeeper	The H.323 gatekeeper to use for outbound H.323 calls from this location. For more information, see About H.323 gatekeepers and SIP proxies.
SNMP NMS	The Network Management System to which SNMP traps for all Conferencing Nodes in this location will be sent. For more information, see Enabling SNMP.
SIP proxy	The SIP proxy to use for outbound SIP calls from this location. For more information, see About H.323 gatekeepers and SIP proxies.
Lync / Skype for Business server	The Lync / Skype for Business server to use for outbound MS-SIP calls from this location. For more information, see About Lync / Skype for Business servers.
TURN server	The TURN server to use when ICE clients (such as Lync / Skype for Business clients and Infinity Connect WebRTC clients) located outside the firewall connect to a Conferencing Node in this location. For more information, see Using TURN servers with Pexip Infinity.
STUN server	The STUN server to use when ICE clients (such as Lync / Skype for Business clients and Infinity Connect WebRTC clients) located outside the firewall connect to a Conferencing Node in this location. For more information, see Using STUN servers with Pexip Infinity.
Client STUN servers	The STUN servers to be used by Infinity Connect WebRTC clients when they connect to a Conferencing Node in this location. Note that **stun.l.google.com** is added by default to every location. For more information, see Using STUN servers with Pexip Infinity.
MTU	(Maximum Transmission Unit) - the size of the largest packet that can be transmitted via the network interface for this system location. Whether you need to specify a value here will depend on the settings of your firewall rules.
DSCP value for media	This is an optional setting used to prioritize different types of traffic in large, complex networks. This DSCP value tags the media traffic from Conferencing Nodes in this system location that is sent line side to endpoints and over the IPsec backplanes to other Pexip Conferencing Nodes.

DSCP value for signaling	This is an optional setting used to prioritize different types of traffic in large, complex networks. This DSCP value tags the signaling traffic from Conferencing Nodes in this system location that is sent line side to endpoints and over the IPsec backplanes to other Pexip Conferencing Nodes. Note that some IPsec traffic between nodes — configuration synchronization and other non-realtime traffic — remains untagged. Also see DSCP value for management traffic in **Global Settings**.
Primary overflow location	An optional field where you can select the system location to handle media when capacity is reached in the current location. For more information, see Location capacity overflow.
Secondary overflow location	An optional field where you can select the system location to handle media when capacity is reached in both the current location and the **Primary overflow location**. If you are using dynamic cloud bursting, your bursting location containing your overflow nodes must be a **Primary overflow location**; the **Secondary overflow location** can only be used for standard overflow i.e. to other "always on" nodes.
Pexip Infinity domain (for Lync / Skype for Business integration)	The name of the SIP domain that is routed from Microsoft Lync / Skype for Business to this Pexip Infinity location, either as a static route or via federation. This is an optional field. If configured, it is used instead of the global Pexip Infinity domain in outbound calls to Lync / Skype for Business from Conferencing Nodes in this location.
Policy profile	The policy profile to be used by Conferencing Nodes in this system location. For more information see **Using external and local policy to control Pexip Infinity behavior**.

About H.323 gatekeepers and SIP proxies

You can configure the Pexip Infinity platform with the addresses of one or more H.323 gatekeepers and SIP proxies. These are the call control systems that can be used to route outbound H.323 and SIP calls on behalf of Pexip Infinity.

Each System location can have a nominated H.323 gatekeeper and/or a SIP proxy — these are used to route outbound H.323 and SIP calls from that location.

Each Call Routing Rule can also have a nominated H.323 gatekeeper and/or a SIP proxy — these are used to route outbound calls for rules that are targeted at external H.323 and SIP systems.

The H.323 gatekeepers and SIP proxies configured here are used for **outbound calls from** Pexip Infinity. They do not determine the systems used to route **inbound calls to** Pexip Infinity. Pexip Infinity **will not** automatically register with any systems configured here. To route inbound calls from a gatekeeper or SIP proxy, you must configure these systems with neighbor zones that direct calls to Pexip Infinity. For more information see Call control.

Outbound calls are made from Pexip Infinity when:

- a conference participant uses an Infinity Connect client to add another participant to the call
- the administrator manually dials out to a participant from a Virtual Meeting Room
- a participant is automatically dialed out to from a Virtual Meeting Room
- a third party uses the API to place a call to a participant
- the Pexip Distributed Gateway is used to interwork a call.

When a Pexip Infinity Connect user who is connected to a particular Conferencing Node adds another participant, the call will be routed to the H.323 gatekeeper or SIP proxy associated with the system location to which that Conferencing Node belongs. If you do not specify an H.323 gatekeeper or SIP proxy for a system location, Pexip Infinity Connect users connected to Conferencing Nodes in that location may still be able to dial in other participants. To enable this, the user must enter an IP address or FQDN (and for the latter, DNS must be configured properly), thus allowing Pexip Infinity to dial the participant directly.

To add, edit or delete an H.323 gatekeeper, go to **Call Control > H.323 Gatekeepers**.

To add, edit or delete a SIP proxy, go to **Call Control > SIP Proxies**.

After adding the details of the H.323 gatekeeper or SIP proxy, you must then add it to the relevant location (**Platform Configuration > Locations**) or Call Routing Rule (**Service Configuration > Call Routing**).

About Lync / Skype for Business servers

Pexip Infinity uses Lync / Skype for Business* servers to route **outbound** MS-SIP calls.

Outbound calls are made from Pexip Infinity when:

- a conference participant uses an Infinity Connect client to add another participant to the call
- the administrator manually dials out to a participant from a Virtual Meeting Room
- a participant is automatically dialed out to from a Virtual Meeting Room
- a third party uses the API to place a call to a participant
- the Pexip Distributed Gateway is used to interwork a call.

You can configure the addresses of one or more Lync/SfB servers within Pexip Infinity. These can be Front End Servers or a Directors; they cannot be Edge Servers.

Within an **on-prem** Lync / Skype for Business environment, you must then associate a specific Lync/SfB server with a System location, Call Routing Rule or Virtual Reception to instruct Pexip Infinity to use that Lync/SfB server when it places an outbound MS-SIP call from a Conferencing Node in that location, or when it matches that routing rule.

Within a **public DMZ** Lync / Skype for Business deployment, you must ensure that a Pexip Infinity location is **not** configured to route calls to a specific Lync/SfB server. This is to ensure that each Conferencing Node uses DNS to locate an appropriate Lync/SfB system via which to route outbound calls.

To add, edit or delete Lync/SfB servers, go to **Call Control > Lync / Skype For Business Servers**.

The available options are:

Option	Description
Name	The name used to refer to this Lync/SfB server in the Pexip Infinity Administrator interface.
Description	An optional description of the Lync/SfB server.
Address	The IP address or FQDN of the Lync/SfB server.
Port	The IP port on the Lync/SfB server to which the Conferencing Node will connect. Default: 5061.
Transport	The IP transport used to connect to the Lync/SfB server. Default: *TLS*.

To associate a Lync/SfB server with a System location, go to **Platform Configuration > Locations**, to associate it with a Call Routing Rule, go to **Service Configuration > Call Routing**, and to associate it with a Virtual Reception that is acting as an IVR gateway to scheduled and ad hoc Lync/SfB meetings, go to **Service Configuration > Virtual Receptions**.

For more information on integrating Pexip Infinity with Microsoft Lync, see Using Microsoft Lync / Skype for Business with Pexip Infinity.

* Note that where this documentation refers to "Lync/SfB", it represents both Microsoft Lync and Skype for Business unless explicitly stated otherwise.

Using TURN servers with Pexip Infinity

TURN servers provide a mechanism for relaying media between devices. ICE (Interactive Connectivity Establishment) is a framework that allows endpoints to discover paths through which they can exchange media — such as via a TURN server — where direct routing between those devices may not be possible, e.g. when a device is on a private network or is behind a NAT.

Pexip Conferencing Nodes can utilize a TURN server and negotiate TURN relays with the following ICE capable clients:

- Lync / Skype for Business clients
- Infinity Connect WebRTC clients (Web App on Chrome, Firefox and Opera, and the mobile and desktop clients)

If these endpoints will be connecting to privately-addressed "on-premises" Conferencing Nodes, you **must** configure Pexip Infinity with the address of at least one TURN server that it can offer to ICE clients.

A TURN server is not required if your Conferencing Nodes are publicly-addressable, either directly or via static NAT. For more information, see When is a reverse proxy, TURN server or STUN server required?.

Note that the H.323 specification has no concept of ICE.

The TURN servers that you use with Pexip Infinity:

- must have a public IP address (located either on the public internet or in a DMZ)
- unless a separate STUN server has also been configured, they must be deployed in such a way that traffic from the Conferencing Node towards the TURN server appears as coming from the Conferencing Node's public NAT address (its server reflexive address), otherwise this will prohibit some Lync / Skype for Business communication scenarios
- must be routable from your Conferencing Nodes
- must be standards-based (supporting **RFC 5766**), for example a VCS Expressway or a Pexip TURN server (see **Pexip Reverse Proxy and TURN Server Deployment Guide**).

When using a TURN server with a Conferencing Node:

- Conferencing Nodes only use TURN over UDP (not TCP). However, Conferencing Nodes will perform ICE TCP negotiation.
- Conferencing Nodes always communicate with a TURN server over a single UDP port (default UDP/3478). UDP media is multiplexed from the Conferencing Node to that single port on the TURN server. The TURN server will reply back to the same port pair on the Conferencing Node. The TURN server never initiates a connection towards a Conferencing Node.

Note that Microsoft A/V Edge Server does not support RFC 5766 and therefore cannot be used as a TURN server with Pexip Infinity.

Nominating the TURN servers used by Pexip Infinity

Within Pexip Infinity you specify one or more TURN servers, and then associate one of those TURN servers with each System location and Call Routing Rule. The same TURN server can be used by more than one location or rule.

- To add, edit or delete TURN server connection details, go to **Call Control > TURN Servers**. The options are:

Option	Description
Name	The name used to refer to this TURN server in the Pexip Infinity Administrator interface.
Description	An optional description of the TURN server.
Address	The IP address of the TURN server.
Port	The IP port on the TURN server to which the Conferencing Node will connect. Default: 3478.
Username Password	The credentials of an account on the TURN server that can be used to create bindings.

- To associate a TURN server with a location, go to **Platform Configuration > Locations** and configure the relevant location.
- To associate a TURN server with a Call Routing Rule, go to **Service Configuration > Call Routing** and configure the relevant rule.

After adding the details of the TURN server, you must add it to the relevant locations and Call Routing Rules.

How Pexip Infinity decides which TURN server to offer

The TURN server offered by Pexip Infinity to ICE clients is determined as follows:

- **Conferences**: uses the TURN server associated with the location of the Conferencing Node that is handling the call signaling.
- **Point-to-point calls via the Pexip Distributed Gateway**: uses the TURN server associated with the Call Routing Rule that matched the call request. If there is no TURN server associated with the rule, then the TURN server associated with the

location of the Conferencing Node that is handling the call is used instead. Note that rules can optionally be configured on a per-location basis.

If a TURN server is not configured for the location or rule, a TURN server relay candidate will not be offered.

When a Conferencing Node receives a call from an ICE client, it sends a request to the TURN server to allocate bindings to be used by that client. The client uses these bindings to route its call media through the firewall to the Conferencing Node. The Conferencing Node allocates up to four TURN bindings per call (made up of two TURN bindings per media stream for both audio and video).

Using STUN servers with Pexip Infinity

A STUN server allows clients, such as Conferencing Nodes or Infinity Connect WebRTC clients, to find out their public NAT address.

When a client is deployed behind a NAT, it can send a STUN request to the STUN server, which responds back to the client and tells it from which IP address it received the STUN request. Using this method, the client can discover its public NAT address, which is important in order for ICE to work between Conferencing Nodes and other ICE-enabled clients (for example, WebRTC and Lync / Skype for Business clients). In relation to ICE, this public NAT address is also known as the server reflexive address.

In Microsoft Lync and Skype for Business deployments it is essential that a Conferencing Node can discover its public NAT address.

Conferencing Nodes

If a Conferencing Node is deployed on a private network behind a NAT, its system location may already be configured with the details of a TURN server (such as the Pexip TURN server). Often, that TURN server can act as a STUN server and a separate STUN server is not normally required.

By default, Conferencing Nodes send their STUN requests to the TURN server, but if the TURN server is not located outside of the enterprise firewall then the Conferencing Node will not be able to discover its public NAT address. If this is the case in your deployment scenario, you must configure a separate STUN server — the Conferencing Node's STUN requests will then be sent to the STUN server, instead of the TURN server.

A STUN server is not required if:

- your Conferencing Nodes are publicly-addressable, either directly or via static NAT, or
- the STUN requests sent from the Conferencing Nodes to the TURN server will return the public NAT address of the Conferencing Node.

The STUN servers that you use with Pexip Infinity must be located outside of the enterprise firewall and must be routable from your Conferencing Nodes.

Infinity Connect WebRTC clients

When connecting to a privately-addressed Conferencing Node, Infinity Connect WebRTC clients that are behind a NAT may also use a STUN server to find out their public NAT address.

When an Infinity Connect WebRTC client connects to a Conferencing Node, the node will provision it with a set of STUN server addresses that it should use to discover its public NAT address.

For more information, see When is a reverse proxy, TURN server or STUN server required?.

How Conferencing Nodes decide which STUN server to use

The STUN server used by a Pexip Infinity Conferencing Node handling a call is determined as follows:

- **Conferences**: uses the STUN server associated with the location of the Conferencing Node that is handling the call signaling.
- **Point-to-point calls via the Pexip Distributed Gateway**: uses the STUN server associated with the Call Routing Rule that matched the call request. If there is no STUN server associated with the rule, then the STUN server associated with the location of the Conferencing Node that is handling the call signaling is used instead. Note that rules can optionally be configured on a per-location basis.

If a STUN server is not configured for a location or rule, but a TURN server is configured, the Conferencing Node will send STUN requests to that TURN server.

Nominating the STUN servers used by Pexip Infinity and Infinity Connect WebRTC clients

Within Pexip Infinity you can configure the addresses of one or more STUN servers. You then associate those STUN servers with each System location (with separate configuration for the STUN server used by Conferencing Nodes in that location, and the STUN servers to offer to Infinity Connect clients connected to that Conferencing Node), and with each Call Routing Rule.

Configuring STUN server addresses

To add, edit or delete STUN server connection details, go to **Call Control > STUN Servers**. The options are:

Option	Description
Name	The name used to refer to this STUN server in the Pexip Infinity Administrator interface.
Description	An optional description of the STUN server.
Address	The IP address or FQDN of the STUN server. This should not be the same address as any of your configured TURN servers.
Port	The IP port on the STUN server to which the Conferencing Node will connect. Default: 3478.

Note that Pexip Infinity ships with one STUN server address already configured by default: **stun.l.google.com**. This STUN server is typically used by Infinity Connect WebRTC clients and uses port 19302 (rather than the common 3478).

Associating STUN server addresses with Conferencing Nodes

To associate a STUN server address with a Conferencing Node, you must configure the node's system location:

1. Go to **Platform Configuration > Locations**.
2. Select the Conferencing Node's location.
3. Select a **STUN server** and select **Save**.

All Conferencing Nodes in that location will use the nominated STUN server for conference calls.

Associating STUN server addresses with gateway calls

If a gateway call is being placed to an ICE-enabled client (such as Lync / Skype for Business clients and Infinity Connect WebRTC clients), the Conferencing Node placing the call will use the STUN server associated with the matching rule. (For gateway calls, the Conferencing Node does not use the STUN sever associated with its system location.)

To associate a STUN server address with a Call Routing Rule:

1. Go to **Service Configuration > Call Routing**.
2. Select the relevant rule.
3. Select a **STUN server** and select **Save**.

Configuring the STUN server addresses provided to Infinity Connect WebRTC clients

To configure the specific STUN server addresses that are provisioned to Infinity Connect WebRTC clients, you must configure the system locations used by the Conferencing Nodes that the clients connect to:

1. Go to **Platform Configuration > Locations**.
2. Select the Conferencing Node's location.
3. Select one or more **Client STUN servers** and select **Save**.
   Note that **stun.l.google.com** is added by default to every location.

When an Infinity Connect WebRTC client connects to a Conferencing Node in that location, the Conferencing Node will provide it with the addresses of the nominated STUN servers. These STUN servers are used by the client to discover its public NAT address.

If no **Client STUN servers** are configured for that node/location, the Infinity Connect client may still be able to communicate by using a TURN relay, if a TURN server is configured on the Conferencing Node, but this may cause delays in setting up media.

For clients on the same network as the Conferencing Nodes, where no NAT is present, users may find that WebRTC call setup time is improved by removing all **Client STUN servers**.

Configuring policy profiles

Policy profiles are used to configure how Pexip Infinity uses external policy and/or local policy to control Pexip Infinity's call policy and routing decisions.

Each policy profile can be used to:

- control which types of data (e.g. service configuration, media location, participants avatars etc.) are managed via policy - either by external policy, or by local policy (where local policy is supported for that data type) or by both external and local policy
- nominate the address of an external policy server to which the external policy API requests are sent
- specify, per data type, the local policy jinja2 script to be executed against that data (currently service configuration and media location data types only).

You can configure Pexip Infinity to use both external and local policy depending on your requirements. When both external and local policy are enabled, external policy is applied first to retrieve the configuration data from the external system, and then local policy is applied to that retrieved data (which can then conditionally modify that data). See **Using external and local policy to control Pexip Infinity behavior** for more information.

Each system location is configured with a policy profile and that profile is then used by all of the Conferencing Nodes in that location whenever they need to retrieve configuration data. This means that you can use the same policy profile in all locations (and thus all Conferencing Nodes), or if required you can configure many different profiles with, for example, different local policy scripts or different external policy server URIs, and then assign different policy profiles to different system locations.

Note that if you are using external policy within a system location, you must ensure that each Conferencing Node in that location is able to reach the nominated policy server.

To configure policy profiles:

1. Go to **Call Control > Policy Profiles**.
2. Select **Add Policy profile** and then configure that profile. The options are:

Option	Description
Name	The name used to refer to this policy profile in the Pexip Infinity Administrator interface.
Description	An optional description of the policy profile.
External policy server	
URL	The URL of the policy server to use for all external policy API requests from this profile, for example **https://policy.example.com/path**. You can only configure one address URL per policy server. If the request is over HTTPS, Pexip Infinity must trust the certificate presented by the policy server.
Username Password	Optional fields where you can specify the credentials required to access the external policy server.
Avatar policy	
Enable external avatar lookup	If enabled, requests are sent to the external policy server to fetch avatar images to represent directory contacts and conference participants.
Service configuration policy	
Enable external service configuration lookup	If enabled, requests are sent to the external policy server to fetch service configuration data (VMRs, Virtual Receptions, Pexip Distributed Gateway calls etc).

Option	Description
Apply local policy	If enabled, the service configuration retrieved from the local database or an external policy server is processed by the local policy script (which may change the service configuration or cause the call to be rejected).
Script	Only applies if **Apply local policy** is selected. Enter a jinja2 script that takes the existing service configuration (if any) and optionally modifies or overrides the service settings.
Media location policy	
Enable external media location lookup	If enabled, requests are sent to the external policy server to fetch the system location to use for media allocation.
Apply local policy	If enabled, the media location configuration retrieved from the local database or an external policy server is processed by the local policy script (which may change the media location configuration).
Script	Only applies if **Apply local policy** is selected. Enter a jinja2 script that takes the existing media location configuration and optionally modifies or overrides the location settings.
Directory	
Enable external directory lookup	If enabled, requests are sent to the external policy server to fetch directory information (that can be used by some Infinity Connect clients to display a phonebook).
Registration requests	
Enable external registration policy	If enabled, requests are sent to the external policy server to determine whether a device alias is allowed to register to a Conferencing Node.

3. Select **Save**.
4. Go to **Platform Configuration > Locations**.
5. Select each location in turn and specify the **Policy profile** that the Conferencing Nodes in that location should use when making policy decisions.

Pexip Infinity license installation and usage

Before you can place calls to Pexip Infinity services (Virtual Meeting Rooms, Virtual Auditoriums, Virtual Receptions and the Pexip Distributed Gateway), you must install a license.

The terms of your Pexip Infinity license can vary according to your licensing agreement. However, your license ultimately permits the use of the Pexip Infinity software, and specifies the total number of **concurrent** calls that can be made to any Pexip Infinity service (Virtual Meeting Rooms, Virtual Auditoriums, Virtual Receptions, Test Call Services or the Pexip Distributed Gateway) across the entire Pexip Infinity deployment. Two types of license can be installed:

- **Audio**: the total number of concurrent audio-only calls (the audio-only participant can also send and receive images and PDF presentation content).
- **Ports**: the total number of concurrent video calls (allows video, audio, screen share, full motion HD presentations, images and PDF content sharing).

Any call licenses installed on Pexip Infinity prior to version 13 are treated as port licenses.

Note that in addition to having sufficient licenses, your deployment must also have sufficient capacity to support the number of calls being placed. For more information, see Capacity planning.

License allocation

When a Conferencing Node receives a request from a participant to join a Virtual Meeting Room, Virtual Auditorium or Virtual Reception, or to interwork a call using the Pexip Distributed Gateway, it contacts the Management Node to see whether there is an appropriate license available. If so, the call is allowed and one concurrent call license is allocated to that participant for the

duration of the call. When the call is terminated, the license is returned to the pool. In general, each participant consumes a single concurrent license, and no participant can consume more than one license at a time. Specifically:

- A **port** license is consumed when making any type of video call (Full HD, HD and SD).
- If your system includes **audio** licenses:
  - a participant making an audio-only call will consume an audio license
  - if a participant attempts to make an audio-only call, but all audio licenses are currently in use, a port license will be used instead (if one is available, otherwise the call will be rejected due to insufficient licenses as normal).
- If your system **only** includes **port** licenses, then any type of call that normally requires an audio license will consume a port license.
- If you attempt to make a call that requires a **port** license and there are no port licenses currently available, but there is an **audio** license available:
  - the call will go ahead as an audio-only call, and consume an **audio** license
  - the participant will see an "insufficient video licenses" screen, and will be seen as an audio-only participant by the other conference participants
  - the call will not automatically escalate to video if a port license becomes available.
- Infinity Connect clients and all other endpoints consume one call license per connection when sending or receiving media (video, audio, screen share, and/or full motion HD presentation). They consume an audio or a port license as appropriate.
- Infinity Connect control-only participants who are only sending slides (images or PDF) or receiving non-HD presentations **do not** consume a license. Control-only participants who are sharing their screen or receiving full motion HD presentations consume a port license.
- Lync / Skype for Business participants (connected directly to a VMR) who are only sending or receiving presentations consume an audio license.
- Gateway calls consume two licenses: one for the inbound leg of the call and one for the outbound leg (each license is audio or port as appropriate).
- A Fusion connection to a Lync / Skype for Business meeting consumes a single port license (regardless of the number of participants connected to the Lync / Skype for Business meeting).

If the Conferencing Node is unable to contact the Management Node, the call will be permitted on the assumption that a license is available. After the Management Node has been out of contact for a grace period of 14 days, all calls will be rejected.

If there are no licenses available, or the existing license is not valid, the participant will be advised that they are not permitted to join the conference along with the reason why, and a corresponding message will appear in the support log.

For more information about license usage, contact your Pexip authorized support representative.

Insufficient licenses

If your Pexip Infinity system reports **insufficient licenses**, this means that there are valid licenses installed on the system, but at the point at which a participant tried to join a conference, all of the existing call licenses were in use. To remedy this, either wait until one or more existing conferences have completed, thus freeing up some call licenses, or add more call licenses to your system.

In some cases your licenses may include an overdraft. This is intended to cover instances where the number of concurrent calls temporarily exceeds the number of available licenses.

Note that if your system includes port licenses and audio licenses, and you attempt to make a call that requires a **port** license and there are no port licenses currently available, but there is an **audio** license available, then the call will go ahead as an audio-only call and consume an **audio** license. See License allocation above for more information.

Invalid license

If your Pexip Infinity reports an **invalid license**, this could mean that:

- the license has not been activated
- the existing license has expired
- the existing license has become corrupt (this could occur, for example, if the Management Node reboots after an upgrade and comes back up on a different physical blade with a new MAC address).

Licenses become valid at 00:00:00 PST on the day they start and expire at 23:59:59 UTC on the day they expire.

Viewing existing licenses and current usage

To see the licenses that are installed on your system, and the number of licenses currently being consumed, go to **Platform Configuration > Licenses**.

The **Licensing** section states how many concurrent licenses are currently being consumed out of the total available, and then lists the licenses that have been activated successfully. You can select a license to see further information, including the maximum number of **Concurrent** calls that can be made.

The **Stored License Request** section lists any licenses that were not activated automatically, and are awaiting manual activation.

Adding licenses

To add a new license:

1. On the Pexip Infinity Administrator interface, go to **Platform Configuration > Licenses**.
2. Select **Add License**.
3. In the **License entitlement key** field, enter the activation key provided by your authorized Pexip representative.
4. Leave the **Manually activate** checkbox clear (unless you have been instructed to select this option by your Pexip authorized support representative).
5. Select **Save**.

Pexip Infinity will automatically generate an XML file containing the license request (for more information, see Content of the license request XML file). It will then attempt to contact the Pexip licensing server and send it this file in order to activate the license. To enable this, you must have an active internet connection from the Management Node, and your firewall must be configured to allow a connection from the Management Node to **pexip.flexnetoperations.com** (64.14.29.85) on HTTPS (port 443).

- If the license is activated successfully, you will be returned to the **Licensing** page and will see the new license under the **Licensing** section.
- If the activation attempt is unsuccessful (for example, if the Management Node was unable to establish a connection to the Pexip licensing server), or you selected **Manually activate**, the license will be saved as a **Stored license request**. You must then activate it manually.

Moving a license (e.g. when redeploying a Management Node)

You may need to move a license from one Management Node to another. You would typically need to do this if you are redeploying a Management Node, such as when setting up a new test or demonstration environment, or changing the Management Node's IP address (see Moving, restoring or changing the IP address of the Management Node).

To move a license between Management Nodes, you must deactivate the existing license on your 'old' Management Node before reactivating it on the 'new' Management Node.

On the **old** Management Node:

1. Go to **Platform Configuration > Licenses**.
2. Select the license you want to deactivate.
3. Make a note of the **License entitlement key**.
4. Select **Return license**.

The system will attempt to contact the Pexip licensing server to automatically deactivate the license:

- If successful, the license will no longer appear in the list of licenses.
- If unsuccessful, you must manually deactivate the license by following the same steps as described in Manually activating a stored license request.

On the **new** Management Node:

1. Go to **Platform Configuration > Licenses**.
2. Select **Add License**.

3. In the **License entitlement key** field, enter the entitlement key from the 'old' Management Node that you noted previously.
4. Select **Save**.

The system will attempt to activate the license on the 'new' Management Node:

- If the license is activated successfully, you will be returned to the **Licensing** page and will see the new license under the **Licensing** section.
- If the activation attempt is unsuccessful (for example, if the Management Node was unable to establish a connection to the Pexip licensing server), or you selected **Manually activate**, the license will be saved as a **Stored license request**. You must then activate it manually.

Manually activating a stored license request

To manually activate a stored license request:

1. Go to **Platform Configuration > Licenses**.
   The **Stored License Request** section lists the licenses that were not able to be activated automatically, and are awaiting manual activation.
2. Select the license you wish to activate.
3. Select **Export stored license request**.
   This generates an XML file containing the request (for more information, see Content of the license request XML file).
4. Download the XML file and send it to your authorized Pexip representative for activation. They will respond with an XML activation file.
5. When you receive the XML file, go back to **Platform Configuration > Licenses**, select the stored license request, and select **Complete stored license request**.
6. Browse to the location of the XML activation file you have received.
7. Select **Save**.

The license should now be activated and appear in the **Licensing** section.

Repairing a license

If a license becomes corrupt (for example if the Management Node MAC address has changed), a **Repair trusted license storage** button will appear under the license information. To reactivate the license, select **Repair trusted license storage**. This will normally resolve the issue automatically; in some circumstances the repair operation will result in a stored license request, in which case follow the procedure above for Manually activating a stored license request.

Content of the license request XML file

During both automatic and manual license activation, the same license request XML file is sent to the Pexip licensing server. The content of this file includes:

- the license request information (i.e. the detail of what is being requested)
- opaque machine identifiers (computed from various information sources, but not exposing the actual values).

Managing TLS and trusted CA certificates

TLS certificates are used by the Management Node and each Conferencing Node to verify their identity to clients connecting to them over HTTPS (web) or SIP TLS. These clients include:

- video endpoints
- web browsers (including Infinity Connect Web App)
- Infinity Connect Mobile clients (certificates are mandatory for these clients)
- third-party video network infrastructure devices.

Wildcard TLS certificates are not supported in SIP or Microsoft Lync / Skype for Business environments (as per RFC 5922). If you are using SIP or Lync / Skype for Business, your Conferencing Nodes must not use wildcard TLS certificates.

Note that communication between the Management Node and Conferencing Nodes, and between Conferencing Nodes themselves, does not rely on TLS certificates; instead it uses an IPsec transport. For more information see Encryption methodologies.

This topic covers:

- Certificate usage overview
- Managing a node's TLS server certificate
- Verifying peer certificates for SIP TLS connections
- Using OCSP to check the status of certificates
- Mutual TLS authentication and client/server certificates
- Managing trusted CA certificates

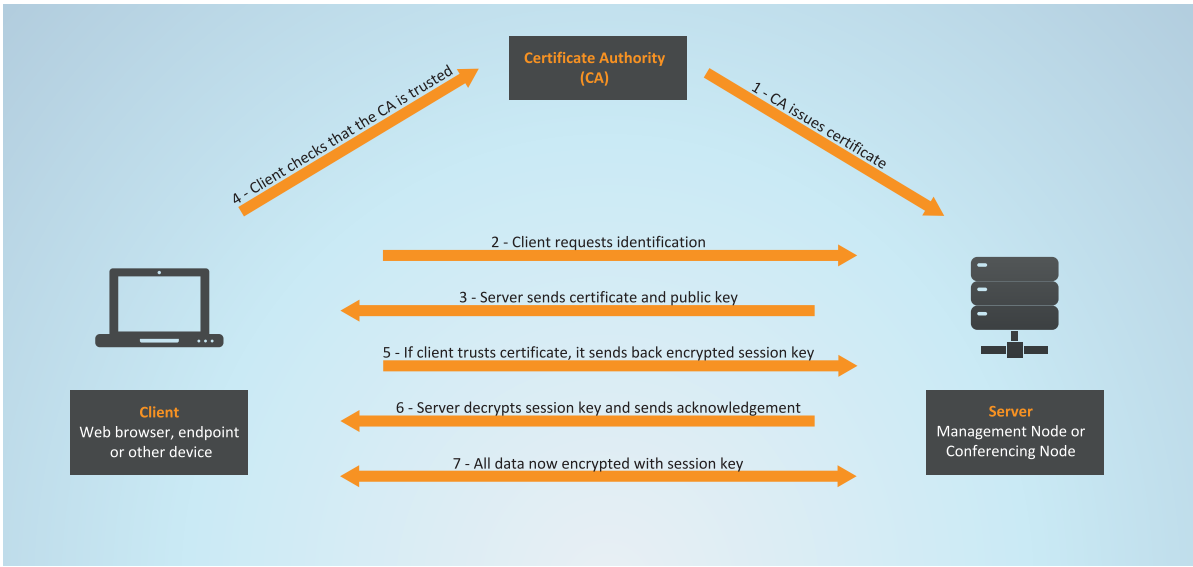
Certificate usage overview

The clients that connect to Pexip Infinity over TLS must trust the identity of the Certificate Authority (CA) that signed the node's TLS certificate. The Pexip Infinity platform ships with a self-signed certificate for the Management Node, and each Conferencing Node is deployed with a self-signed certificate. These certificates have a 4096 bit public key and are also appended with 2048 bit Diffie-Hellman parameters.

As these certificates are self-signed, they will not be trusted by clients. We therefore recommend that you replace these certificates with your own certificates that have been signed by either an external CA or a trusted internal CA. You can use Pexip Infinity's inbuilt Certificate Signing Request (CSR) generator to assist in acquiring a server certificate from a Certificate Authority.

You can use a tool such as https://www.sslshopper.com/ssl-checker.html to verify certificates and the chain of trust (specify port 5061 i.e. use the format **<domain>:5061** for the server hostname to ensure that SIP TLS connections are checked).

The Management Node and Conferencing Nodes enable HSTS (HTTP Strict Transport Security) to ensure greater security. This means that if your deployment moves from using a valid TLS certificate to using an invalid certificate (e.g. you redeploy a Conferencing Node, or your certificate expires or is invalidated for some reason) then certain web browsers will stop you from accessing that node via the web when using the DNS name of that node, until you correct the certificate issue. You may browse directly to the IP address of the node in the meantime.



Overview of process for encrypting communication using TLS

In general, to achieve encrypted communication using TLS the following must happen:

1. The CA issues a signed certificate which is uploaded to the server.
2. When a client needs to communicate with the server, it sends a request to the server asking it to provide identification.
3. The server sends back a copy of its TLS certificate and its public key.

4. The client checks whether the CA that issued the certificate is one that it trusts.
5. If the CA is trusted, and if the certificate is otherwise valid, the client creates a session key encrypted with the server's public key and sends it to the server.
6. The server decrypts the session key. It then uses the session key to encrypt an acknowledgment which it sends to the client in order to initiate the encrypted communication.
7. The server and the client now encrypt all communication using the session key.

Alarms

An alarm is raised on the Management Node if:

- the Management Node or a Conferencing Node has no associated TLS certificate
- a TLS certificate has an incomplete chain of trust to the root CA certificate
- one or more of your trusted CA certificates is due to expire within the next 30 days.

Managing a node's TLS server certificate

You can upload and view the TLS server certificates that are used by the Management Node and by each Conferencing Node.

Note that after making any changes to certificates, there will be a delay of up to 1 minute while the files are synchronized to the relevant Management Node or Conferencing Node.

Uploading a TLS server certificate

To upload a new TLS server certificate for the Management Node or a Conferencing Node:

1. From the Pexip Infinity Administrator interface, go to **Platform Configuration > TLS Certificates**.
2. Select **Add TLS certificate**.
3. Complete the following fields:

TLS certificate	Paste the PEM-formatted certificate into the text area or alternatively select the file containing the new TLS certificate. You must upload the certificate file that you have obtained from the Certificate Authority (typically with a .CRT or .PEM extension). Do not upload your certificate signing request (.CSR file). The certificate must be valid for the hostname or FQDN of the Management Node or Conferencing Node to which it will be assigned. You can paste multiple certificates into the text area, but one of those certificates must pair with the associated private key.
Private key	Paste the PEM-formatted private key into the text area or alternatively select the file containing the private key that is associated with the new TLS certificate. Private key files typically have a .KEY or .PEM extension. Pexip Infinity supports RSA and ECDSA keys.
TLS parameters	Optionally, paste any additional PEM-formatted parameters into the text area or alternatively select the file containing the parameters that are to be associated with the new TLS certificate. Custom DH parameters and an EC curve name for ephemeral keys can be added. Such parameters can be generated through the OpenSSL toolkit using the commands `openssl dhparam` and `openssl ecparam`. For example, the command `openssl dhparam -2 -outform PEM 2048` generates 2048 bit DH parameters. Note that these parameters can alternatively be added 'as is' to the end of the TLS certificate.
Nodes	Select one or more nodes to which the new TLS certificate is to be applied. If required, you can upload a certificate and then apply it to a node later.

4. Select **Save**.

Viewing or modifying existing TLS certificates and changing node assignments

To view information about an existing TLS certificate, change a certificate's contents, or change the nodes to which a certificate is applied:

1. Go to **Platform Configuration > TLS Certificates**.
   By default you are shown a list and the current status of **All certificates** that have been uploaded. A **Status** of ***Temporary*** means that it is a self-signed certificate.
   You can alternatively select to view **Certificates by Node** to see which certificate has been assigned to the Management Node or to a particular Conferencing Node.
2. Select the subject name of the certificate you want to view or modify.
   The decoded certificate data is shown, including any chain of trust information.
3. If required, you can modify the:
   - **Nodes** to which the certificate is assigned
   - **TLS certificate** data
   - **TLS parameters** associated with the certificate.

   You cannot modify the private key.
4. Select **Save**.

Uploading multiple TLS certificate files

To upload a batch of TLS certificates:

1. Go to **Platform Configuration > TLS Certificates**.
2. Select **Import files**.
3. Select **Choose Files** to pick one or more PEM-formatted text files that you want to import.
   - The files should contain server TLS certificates with matching private keys.
   - Private keys can be uploaded as separate files or appended to the server TLS certificate file(s).
   - DH or EC parameters may be appended to each server TLS certificate.

   Note that trusted CA certificate files can also be imported via this method if required.
4. Select **Import**.
   This adds the certificates in the selected files to the existing list of TLS certificates (or to the list of trusted CA certificates, if appropriate). If a certificate with the same subject name already exists (e.g. when replacing an expired certificate), that certificate will be updated with the new contents from the file. Any duplicate certificates are ignored.
5. You can then select each imported TLS certificate in turn and assign it to a Management Node or one or more Conferencing Nodes as appropriate.

Verifying peer certificates for SIP TLS connections

For connections over SIP TLS, you can use the **SIP TLS verification mode** setting to control whether the certificate presented by the peer is verified before the connection is allowed. When this setting is enabled:

- The peer certificate is verified (in date, issued by a trusted authority etc).
- OCSP, if enabled, is used to check that the certificate has not been revoked (see Using OCSP to check the status of certificates below for more information).
- The identity of the peer as presented in the certificate is checked against the identity expected by Pexip Infinity.
- The peer certificate must be a **client** certificate (see Mutual TLS authentication and client/server certificates below for more information) — otherwise you will get "unsupported certificate" errors in the support log.

When this setting is enabled, all peers (including endpoints connecting directly with Pexip Infinity) must have their own certificate.

To enable or disable this setting, go to **Platform Configuration > Global Settings**. The options available are:

Field	Description
SIP TLS verification mode	Determines whether to verify the peer certificate for connections over SIP TLS. ***Off***: the peer certificate is not verified; all connections are allowed. ***On***: the peer certificate is verified, and the peer's remote identities (according to RFC5922) are compared against the Application Unique String (AUS) identified by Pexip Infinity before the connection is allowed.

Using OCSP to check the status of certificates

Pexip Infinity allows you to use Online Certificate Status Protocol (OCSP) to check whether a certificate has been revoked. TLS certificates that support OCSP are encoded with the URL of an OCSP responder — a server that will check and respond with the status of the certificate.

OCSP checking only applies when **SIP TLS verification mode** is ***On***. To enable or disable the use of OCSP, and to configure the URL of the OCSP responder, go to **Platform Configuration > Global Settings**. The options available are:

Field	Description
OCSP state	Determines whether OCSP is used to check the status of TLS certificates. ***Off***: OCSP is not used. ***On***: OCSP is used, and the request is sent to the URL specified in the TLS certificate. If no URL is specified in the TLS certificate, the **OCSP responder URL** configured below is used. ***Override***: OCSP is used, and the request is sent to the **OCSP responder URL** configured below, regardless of any URL encoded in the TLS certificate.
OCSP responder URL	The URL to which OCSP requests are sent if either: • the **OCSP state** is set to ***On*** but no URL is present in the TLS certificate, or • the **OCSP state** is set to ***Override*** (in which case any URL present in the certificate is ignored).

Mutual TLS authentication and client/server certificates

If a Conferencing Node makes an outbound connection to, or receives an incoming connection from, a video network infrastructure device (such as a Cisco VCS) that is configured to perform TLS verification checking, then that external system will verify that the certificate on the Conferencing Node is valid. In these cases, both the Conferencing Node and the external system can adopt the role of a client as well as acting as a server. Therefore, for mutual TLS authentication — where both parties are verifying the other party's certificate — both the Conferencing Node's TLS server certificate and the external system's certificate must be capable of being used as a **client** certificate as well as a **server** certificate.

A certificate's capabilities are contained in its Enhanced Key Usage properties. They indicate if the certificate can be used for server authentication (typically shown as "TLS Web Server Authentication") and for client authentication (typically shown as "TLS Web Client Authentication").

When requesting certificates for your Conferencing Nodes, if you want the node to be able to verify itself as a client when connecting to an external system, then you must request that the certificate can act as a client certificate (as well as a server certificate). All CSRs generated via Pexip Infinity's inbuilt CSR generator always request client certificate and server certificate capabilities.

If you create your certificate signing requests via the OpenSSL toolkit, then you can modify the `[v3_req]` section of your openssl request configuration file so that it contains `extendedKeyUsage=serverAuth, clientAuth`. Other Certificate Authorities have different methods of requesting client authentication.

The following table summarizes the certificate requirements for inbound connections to, and outbound connections from Conferencing Nodes:

	Inbound SIP TLS connections	Outbound SIP TLS connections

When **SIP TLS verification mode** is ***Off***	• The Conferencing Node's certificate must have **server** authentication properties.	• Pexip Infinity enables ADH ciphers (Anonymous Diffie-Hellman) — this enables support for endpoints that do not have certificates.
		• Pexip Infinity may or may not authenticate the far end, depending on which cipher suite is selected by the far end:
		  ○ if an ADH cipher is selected, the far end is not authenticated and certificates are not exchanged
		  ○ if a cipher that requires authentication is selected and a certificate is exchanged, that certificate must be valid (issued by a trusted authority, in date etc.) and it must be a **server** certificate.

When **SIP TLS verification mode** is ***On***	As above, plus: • The client system/endpoint must present a valid certificate (issued by a trusted authority, in date etc.) and it must be a **client** certificate. • The identity of the peer as presented in the certificate must match the identity expected by Pexip Infinity.	As above, plus: • The identity of the peer as presented in the certificate must match the identity expected by Pexip Infinity. This means that Pexip Infinity cannot connect out over TLS to endpoints that don't have a certificate (even if an ADH cipher is selected) — you must use TCP/UDP instead.
When **OCSP state** is ***On*** or ***Override*** (and **SIP TLS verification mode** is ***On***)	• The client certificate must not be revoked.	• The far end certificate must not be revoked.
If the far end system performs a TLS verification process	• The Conferencing Node's certificate must have **client** authentication properties and must **not** be a self-signed certificate.	• The Conferencing Node's certificate must have **client** authentication properties and must **not** be a self-signed certificate.

Managing trusted CA certificates

Trusted CA certificates are used within Pexip Infinity to:

- verify client certificates presented to Pexip Infinity when SIP TLS verification mode is enabled
- provide a certificate chain of trust when clients connect to a Conferencing Node over SIP TLS
- verify the server certificate on a video network infrastructure device when a Conferencing Node makes an SIP TLS outbound connection to that device, and that device chooses a cipher suite that requires authentication and a certificate is exchanged
- verify connections to an LDAP server.

Pexip Infinity ships with an inbuilt list of trusted CA certificates. This list is based on the Mozilla CA Certificate Store and cannot be modified.

In addition, you can upload a customized set of trusted CA certificates to the Pexip Infinity platform.

Chain of trust

For a server's TLS certificate to be trusted by a client, the client must be configured to trust the Certificate Authority (CA) that signed the server certificate. Many CAs do not sign with their root certificate, but instead with an intermediate certificate. Clients, however, may only trust the root CA. Therefore the server (in this case the Management Node or Conferencing Node) is often required to present a full certificate chain along with their TLS server certificate.

When this is the case, the chain of intermediate CA certificates must be installed on the Management Node to ensure that the certificate chain of trust is properly established when clients connect to a Conferencing Node over SIP TLS.

To do this you must upload a custom **Trusted CA certificates** file that contains all the required CA certificates, one after each other, in PEM format.

Uploading and managing additional trusted CA certificates

You can upload a customized set of trusted CA certificates to the Pexip Infinity platform. Any trusted CA certificates uploaded here are used in addition to the default set of trusted CA certificates that ships with Pexip Infinity.

To manage the set of custom trusted CA certificates, go to **Platform Configuration > Trusted CA Certificates**. This shows a list and the current status of all the trusted CA certificates that have been uploaded. From here you can:

- **Upload a file of Trusted CA certificates**: select **Import files**, select **Choose Files** to pick one or more PEM files that you want to import, and then select **Import**.

  This adds the certificates in the selected files to the existing list of trusted CA certificates (or to the list of TLS certificates, depending on the certificate types contained in the file). If a certificate with the same subject name already exists (e.g. when replacing an expired certificate), that certificate will be updated with the new contents from the file. Any duplicate certificates are ignored.
- **View or modify an existing certificate**: select the **Subject name** of the certificate you want to view. The decoded certificate data is shown.

  If required, you can modify the PEM-formatted certificate data and select **Save**.
- **Download all certificates**: select **Export**. A **ca-certificates.pem** file containing all of the custom-added certificates in PEM format is created and automatically saved to your local file system.
- **Delete one or more certificates**: select the boxes next to the certificates to be deleted, and from the **Action** drop-down menu select ***Delete selected Trusted CA certificates*** and select **Go**.

Certificate signing requests (CSRs)

To acquire a server certificate from a Certificate Authority (CA), a certificate signing request (CSR) has to be created and submitted to the CA. You can generate a CSR from within Pexip Infinity, and then upload the returned certificate associated with that request.

CSRs generated via Pexip Infinity always request client certificate and server certificate capabilities.

Creating a certificate signing request

To generate a CSR within Pexip Infinity:

1. Go to **Utilities > Certificate Signing Requests**.
2. Select **Add Certificate signing request**.
3. Complete the following fields:

Subject name	Select the name to be specified as the Common Name field of the requested certificate's subject. This is typically set to the FQDN of the node on which the certificate is to be installed. The available options are prepopulated with the FQDNs (hostname plus domain) of the Management Node and each currently deployed Conferencing Node. The list also includes any **SIP TLS FQDN** names of your Conferencing Nodes, if such names have been configured and are different from the node's FQDN. If you want to specify a custom Common Name instead, select ***User-provided custom Common Name***.
Custom subject name	Enter the name that you want to use as the Common Name field of the requested certificate's subject, if you have selected ***User-provided custom Common Name*** above.
Private key type	Select the type of private key to generate, or select ***Upload user-provided private key*** if you want to provide your own private key. Default: RSA (2048bit)
Private key	Only applies if you have selected ***Upload user-provided private key*** above. Enter the PEM formatted RSA or ECC private key to use when generating your CSR. You can either paste the key into the input field or upload the private key file from your local file system.

Subject alternative names	Select the subject alternative names to be included in the CSR. This allows the certificate to be used to secure a server with multiple names (such as a different DNS name), or to secure multiple servers using the same certificate. You can choose from the same list of names presented in the **Subject name** field (the name you have already chosen as the Common Name is by default automatically included in the Subject alternative names list). In some deployments it may be more practical to generate single CSR in which all of your Conferencing Node FQDNs are included in the list of subject alternative names. This means that the same single server certificate returned by the CA can then be assigned to every Conferencing Node. When integrating with Microsoft Lync / Skype for Business, subject alternative name entries must be included for every individual Conferencing Node in the public DMZ (public DMZ deployments) or in the trusted application pool (on-prem deployments). See **Certificate creation and requirements for Microsoft Lync / Skype for Business integrations** more information.
Additional subject alternative names	Optionally, enter a comma-separated list of additional subject alternative names to include in the CSR. For example, when integrating with on-prem Lync / Skype for Business deployments you would typically need to add the trusted application pool FQDN.
Additional subject fields (if required you can enter the following additional CSR attributes; these are all blank by default)	
Organization name	The name of your organization.
Department	The department within your organization.
City	The city where your organization is located.
State or Province	The state or province where your organization is located.
Country	The 2 letter code of the country where your organization is located.
Advanced (in most scenarios you should leave the advanced options to their default settings)	
Include Microsoft certificate template extension	Select this option to specify a (Microsoft-specific) certificate template in the CSR. This is needed when using the Certification Authority MMC snap-in to request a certificate from an enterprise CA. Selecting this option causes the 'WebServer' certificate template to be specified. Default: disabled.
Include Common Name in Subject Alternative Names	Specifies whether to include the requested subject Common Name in the Subject Alternative Name field of the CSR. Default: enabled.

4. Select **Save**.
   You are returned to the list of certificate signing requests.
5. Select the CSR you have just created.
   You are shown the decoded certificate data.
6. Scroll to the bottom of the page and select **Download**.
   This downloads the CSR to your local file system, with a filename in the format **<subject-name>.csr**.
   Note that the private key is not downloaded, or included within the CSR.
7. You can now submit this CSR file to your chosen CA for signing.
   The CA will then send you a signed certificate which you can upload into Pexip Infinity (see below).

Uploading the signed certificate associated with a certificate signing request

When the Certificate Authority sends you a signed certificate in response to your CSR, you can upload that certificate into Pexip Infinity and assign it to one or more of your nodes. Make sure that you upload it via the **Certificate Signing Requests** page as this ensures that it is linked with the private key associated with your original CSR.

To upload the signed certificate:

1. Go to **Utilities > Certificate Signing Requests**.
2. Select the original CSR that is associated with the signed certificate.
   You are shown the decoded certificate data.
3. Scroll down the page and in the **Certificate** field either paste the PEM-formatted certificate into the input field or upload the certificate file from your local file system.
   The certificate file that you have obtained from the Certificate Authority typically has a .CRT or .PEM extension. Do not upload your certificate signing request (.CSR file).
4. Select **Complete**.
   Providing it is a valid certificate and is based on the original CSR:
   - the certificate is uploaded and automatically linked with the private key associated with your original CSR
   - the original CSR is deleted
   - you are taken to the **Change TLS Certificate** page.
5. You can now assign that certificate to the Management Node or one of more Conferencing Nodes as required:
   a. From within the **Change TLS Certificate** page go to the **Nodes** field and from the **Available Nodes** list, select the nodes to which you want to assign the certificate and move them into the **Chosen Nodes** list.
   b. Go to the bottom of the page and select **Save**.

For more information about assigning certificates to nodes, see Viewing or modifying existing TLS certificates and changing node assignments.

Troubleshooting

If you receive an error message "Certificate and private key do not appear to be part of the same key pair" when attempting to upload a signed certificate, this most likely means that you have tried to upload the certificate against the wrong CSR.

Modifying a CSR

After a CSR has been created it cannot be modified — the only available actions are to download it (for sending to a CA), or to apply the returned, signed certificate that is associated with that request.

If you need to change the content of a CSR, you should delete the original CSR and create a new CSR with the correct content.

Note that a CSR is automatically deleted when the resulting signed certificate is uploaded.

Integrating with external systems

In many video conferencing deployments, Pexip Infinity will be integrated with other third-party systems and products. An example is Cisco Unified Communications Manager (CUCM), which can be integrated with Pexip Infinity so that when users initiate the **Ad Hoc Conferencing** feature on supported endpoints, the resulting conference will be hosted on Pexip Infinity.

This type of integration requires that communications from the external system are authenticated by Pexip Infinity. (Note that the credentials for this authentication are distinct from those used to access the Pexip Infinity management API.)

Some external systems (such as CUCM 8.6.2) only support access over HTTP rather than the secure HTTPS protocol, so you have the option to enable this if required.

To configure Pexip Infinity with the authentication credentials and protocol to be used by systems which do not use the management API, go to **Platform Configuration > Global Settings**. The options available are:

Field	Description
Enable HTTP access for external systems	Access for external systems is over HTTPS by default. If this box is selected, access over HTTP is also permitted.
External system username	The username presented to Pexip Infinity by external systems attempting to authenticate with it.
External system password	The password presented to Pexip Infinity by external systems attempting to authenticate with it.

Enabling and disabling SIP, H.323, WebRTC and RTMP

All call protocols, except for SIP over UDP, are enabled by default in your Pexip Infinity deployment.

If your deployment does not include any endpoints that support a particular protocol, for security reasons you may want to disable support for those protocols across your entire Pexip Infinity deployment.

SIP over UDP is disabled by default so as to reduce the impact of SIP spam which typically uses UDP.

The configurable protocols are:

- SIP over TCP and TLS, including MS-SIP (Lync / Skype for Business)
- SIP over UDP (for incoming calls)
- H.323
- WebRTC
- RTMP

WebRTC calls can originate from the Infinity Connect desktop client, the Infinity Connect Web App via Google Chrome, Microsoft Edge, Firefox and Opera browsers, and the Infinity Connect Mobile client.

RTMP calls can originate from the Infinity Connect Web App via Internet Explorer and Safari browsers. The Web App will attempt an encrypted RTMPS connection first. For a secure RTMP connection to be established, the **SIP TLS FQDN** must be configured on the Conferencing Node (via **Platform Configuration > Conferencing Nodes**) and it must match the Common Name of its TLS server certificate. If RTMPS fails, it will use an unencrypted connection for media.

RTMP is also used when sending conference content to streaming and recording services.

To enable or disable a particular call protocol across your Pexip Infinity deployment:

1. Go to **Platform Configuration > Global Settings**.
2. Select or clear the following checkboxes, as appropriate:
   - **Enable SIP** — this affects both SIP and MS-SIP (Lync / Skype for Business) over TCP and TLS
   - **Enable SIP UDP** — this affects incoming calls only over SIP UDP
   - **Enable H.323**
   - **Enable WebRTC**
   - **Enable RTMP**
3. After modifying any of these values, each Conferencing Node must be rebooted before the changes will take effect. (It will, however, have an immediate effect on outbound calls placed via the modified protocols.)

Enabling Full HD (1080p)

By default, Pexip Infinity deployments have Full HD (1080p) disabled. This means that endpoints capable of sending Full HD will only be permitted to use HD (720p) resolution.

You can enable Full HD in your deployment, which allows any endpoint capable of Full HD to send video and presentations at 1080p. However, be aware that enabling Full HD will have implications on bandwidth and capacity across your deployment, specifically:

- Full HD calls require approximately double the Conferencing Node resources and double the bandwidth of an HD call.
- With Full HD enabled, all backplane reservations consume resources equivalent to a Full HD call, approximately twice that of an HD call. For more information, see Backplane reservation.

Note that a Full HD call only consumes a single call license (as does any other type of call).

When you enable or disable Full HD, it takes effect for any new conferences initiated after the change has been made.

If you enable Full HD, you may wish to consider applying bandwidth limitations to individual services if appropriate, to enable you to manage your bandwidth usage and server capacity.

To enable or disable Full HD:

1. Go to **Platform Configuration > Global Settings**.
2. From within the **Connectivity** section, deselect or select **Enable 1080p (Full HD)**.

Managing SIP credentials

When sending a SIP request, Pexip Infinity may be challenged for authentication credentials by the host system or proxy.

The authentication challenge will include a realm that identifies the credentials that Pexip Infinity should use to respond to the challenge. The credentials to use will be provided separately by the administrator of the host system or proxy.

All realms, usernames and passwords are case sensitive.

To configure the SIP credentials used by Pexip Infinity for each realm, go to **Call Control > SIP Credentials**. The options are:

Option	Description
Realm	The realm defines the protection space of the host or proxy (such as its name or domain, e.g. sipproxy.example.com) that is challenging Pexip Infinity for authentication.
Username and Password	The username and password presented by Pexip Infinity when responding to the authentication challenge for the given realm.

Conferencing Node configuration

Deploying new Conferencing Nodes

Conferencing Nodes are virtual machines that provide the capacity for conferences. They handle all conference media and signaling.

There is no limit on the number of Conferencing Nodes that you can add to the Pexip Infinity platform. However, each Conferencing Node must have a unique:

- name
- IP address
- hostname
- DNS name (hostname and domain)

For secure deployments, you should also:

- specify a SIP TLS FQDN
- replace the self-signed TLS certificates with your own TLS certificates

Conferencing Nodes can be deployed with dual network interfaces (NICs). Note that you must specify both interface addresses when initially deploying the Conferencing Node; you may also need to assign a static route while deploying the node. For more information, see Conferencing Nodes with dual network interfaces (NICs).

After deploying a new Conferencing Node, it takes approximately 5 minutes before the node is available for conference hosting and for its status to be updated on the Management Node. (Until it is available, the Management Node will report the status of the Conferencing Node as having a last contacted and last updated date of "Never".)

All Conferencing Nodes have identical service configuration, which is obtained automatically from the Management Node.

Do not use VMware, Hyper-V or any other tools to clone instances of existing Conferencing Node virtual machines (VMs). Conferencing Nodes must always be created using the Pexip Infinity Administrator interface or management API.

Prerequisites

- All host servers **must** be synchronized with accurate time before you install the Pexip Infinity Management Node or Conferencing Nodes on them.
- You must enable NTP on the Pexip Infinity Management Node before you deploy any Conferencing Nodes.

Deployment types

To create a new Conferencing Node, first a generic instance of a Conferencing Node VM is created, and then it is configured with the details of the specific Conferencing Node being deployed.

This process has been automated (or partly automated) for environments using VMware ESXi, Microsoft Hyper-V, KVM or Xen hypervisors, which are supported in Pexip Infinity. However, Pexip also provide the Conferencing Node VM template and the ability to generate a Conferencing Node-specific configuration file for deployment in other environments using non-supported hypervisors or orchestration layers.

The different deployment options are described in the table below. A Pexip Infinity deployment can contain Conferencing Nodes created using any combination of these methods.

Deployment type	Description
Automatic (ESXi 4.1, 5.x and 6.0)	Pexip Infinity deploys the Conferencing Node VM on a host server running VMware vSphere ESXi 4.1 or higher. Choose this option if you want to deploy the Conferencing Node immediately, and there is network connectivity to the host server on which the Conferencing Node is to be deployed. For more information, see Automatically deploying a new Conferencing Node on a VMware host.

Deployment type	Description
Manual (ESXi 6.0) Manual (ESXi 5.x) Manual (ESXi 4.1)	Pexip Infinity generates an **.ova** file that you must then manually deploy from within VMware on to an ESXi host in order to create the Conferencing Node VM. Choose one of these options (selecting the appropriate ESXi host version for your environment) if the Management Node does not have existing connectivity to the host server on which the Conferencing Node is to be deployed, or you do not want to deploy the Conferencing Node immediately. For more information, see Manually deploying a Conferencing Node on an ESXi host.
Manual (Hyper-V 2012)	Pexip Infinity generates a file that you must then manually deploy on a host server running either Microsoft Hyper-V Server 2012 and higher, or Windows Server 2012 and higher, in order to create the Conferencing Node VM. Choose this option for all deployments that use Hyper-V. For more information, see Manually deploying a Conferencing Node on a Hyper-V host.
Manual (KVM)	Choose this option to generate an **.ova** file that is suitable for deploying on a KVM host in order to create the Conferencing Node VM. For more information, see Manually deploying a Conferencing Node on a KVM host.
Manual (Xen)	Choose this option to generate an **.ova** file that is suitable for deploying on a Xen host in order to create the Conferencing Node VM. For more information, see Manually deploying a Conferencing Node on a Xen host.
Generic (configuration-only)	Pexip Infinity generates a file containing the configuration of the Conferencing Node. You then upload this file to a generic Conferencing Node that has been created from the Pexip-supplied VM template, in order to configure it with the appropriate settings. Choose this option for any deployments that do not use ESXi, Hyper-V, KVM or Xen hypervisors, or for Hyper-V in a cloud-based environment. For more information, see Deploying a Conferencing Node using a generic VM template and configuration file.

Automatically deploying a new Conferencing Node on a VMware host

To automatically deploy a new Conferencing Node on to a VMware ESXi host:

1. Ensure that the Management Node can connect directly to the ESXi host server over HTTPS (port 443/TCP), even if you are using vCenter server.
2. Go to **Platform Configuration > Conferencing Nodes** and select **Add Conferencing Node**.
3. From the **Deployment type** field, select ***Automatic (ESXi 4.1, 5.x and 6.0)***.
4. Select **Next**.
5. You are now asked to log in to the VM manager via which the Conferencing Node virtual machine is to be created, by completing the following fields:

Option	Description
VM manager	Select the VM manager via which this Conferencing Node VM will be created. If the VM manager does not appear in the drop-down list, you can add it by clicking ✚ to the right of the field.
Username Password	Enter a valid username and password to log in to the selected VM manager.
Verify TLS certificates	Determines whether the certificate presented by the VM manager is verified before the connection is allowed.

6. Select **Next**.
   There may be a slight delay while Pexip Infinity locates and logs in to the VM manager, and obtains information from it.

7. You are now asked to select the datacenter on the VM manager where the Conferencing Node VM will be created, by completing the following field:

Option	Description
Datacenter	Select the path of the datacenter in which the Conferencing Node will be created. The options available in the drop-down menu reflect what is currently configured in the VM manager.

   Note that this step is skipped if there is only one datacenter configured in the VM manager.
8. Select **Next**.
9. You are now asked to nominate the resource pool on the VM manager where the Conferencing Node VM will be created, by completing the following field:

Option	Description
Resource path	Select the path of the resource pool in which the Conferencing Node will be created. The options available in the drop-down menu are filtered by the datacenter selected in the previous step and reflect what is currently configured in the VM manager. To add a new path, you must do so using vCenter Server (or the ESXi host if you are not using vCenter Server) prior to creating the Conferencing Node.

10. Select **Next**.
11. You are now asked to provide the VM manager with parameters relating to the Conferencing Node, by completing the following fields:

Option	Description
Host network	Select the host network on which this Conferencing Node will be deployed. The options shown here are the vSphere switches that have been configured on the VM manager on which this Conferencing Node is being deployed.
Secondary interface host network	If a secondary network interface for this Conferencing Node is required, select the host network to use for that second interface.
Host datastore	Select the datastore on the host server to be used by this Conferencing Node. The options shown here are the datastores that have been configured on the VM manager via which this Conferencing Node is being deployed.

12. Select **Next**.
13. You are now asked to provide the VM manager with information regarding the CPUs and memory of the Conferencing Node, by completing the following fields:

Option	Description
Number of virtual CPUs to assign	Enter the number of virtual CPUs to assign to the Conferencing Node. We recommend no more than one virtual CPU per physical core, unless you are making use of CPUs that support hyperthreading.
System memory (in megabytes) to assign	Enter the amount of RAM (in megabytes) to assign to the Conferencing Node. The number entered must be a multiple of 4. We recommend 1024 MB (1 GB) RAM for each virtual CPU.

14. Select **Next**.
15. You are now asked to provide the network configuration to be applied to the Conferencing Node, by completing the following fields:

Option	Description
Name	Enter the name that will be used to refer to this Conferencing Node in the Pexip Infinity Administrator interface.

Option	Description
Description	An optional field where you can provide more information about the Conferencing Node.
Hostname Domain	Enter the hostname and domain to be assigned to this Conferencing Node. Each Conferencing Node and Management Node must have a unique hostname. The Hostname and Domain together make up the Conferencing Node's DNS name or FQDN. We recommend you assign valid DNS names to all your Conferencing Nodes. For more information, see Assigning hostnames and FQDNs.
IPv4 address	Enter the IP address to be assigned to this Conferencing Node when it is created.
Network mask	Enter the IP network mask to be assigned to this Conferencing Node.
Gateway IPv4 address	Enter the IP address of the default gateway to be assigned to this Conferencing Node.
Secondary interface IPv4 address	The optional secondary interface IPv4 address for this Conferencing Node. If configured, this interface is used for signaling and media communications to clients, and the primary interface is used for communication with the Management Node and other Conferencing Nodes. For more information, see Conferencing Nodes with dual network interfaces (NICs).
Secondary interface network mask	The optional secondary interface network mask for this Conferencing Node.
System location	Select the physical location of this Conferencing Node. If the system location does not already exist, you can create a new one here by clicking ✚ to the right of the field. This will open up a new window showing the **Add System Location** page. For further information see About system locations.
SIP TLS FQDN	A unique identity for this Conferencing Node, used in signaling SIP TLS Contact addresses. For more information, see SIP TLS FQDN.
TLS certificate	The TLS certificate to use on this node. This must be a certificate that contains the above **SIP TLS FQDN**. Each certificate is shown in the format **<subject name> (<issuer>)**.
IPv6 address	The IPv6 address for this Conferencing Node. Each Conferencing Node must have a unique IPv6 address.
Gateway IPv6 address	The IPv6 address of the default gateway.
IPv4 static NAT address	The public IPv4 address used by this Conferencing Node when it is located behind a NAT device. Note that if you are using NAT, you must also configure your NAT device to route the Conferencing Node's **IPv4 static NAT address** to its **IPv4 address**. For more information, see Configuring Pexip Infinity nodes to work behind a static NAT device.
Static routes	From the list of **Available Static routes**, select the routes to assign to the node, and then use the right arrow to move the selected routes into the **Chosen Static routes** list. For more information, see Managing static routes.
Enable distributed database	This should usually be enabled (checked) for all Conferencing Nodes that are expected to be "always on", and disabled (unchecked) for nodes that are expected to only be powered on some of the time (e.g. nodes that are likely to only be operational during peak times).
SSH password	Enter the password to be used when logging in to this Conferencing Node's Linux operating system over SSH. The username will always be ***admin***. Logging in to the operating system is required when changing passwords or for diagnostic purposes only, and should generally be done under the guidance of your Pexip authorized support representative. In particular, do not change any configuration using SSH — all changes should be made using the Pexip Infinity Administrator interface.

16. Select **Finish**.

You will be taken to the **Deploying Conferencing Node** page. The deployment will take several minutes. When complete, the progress bar will show that the Conferencing Node VM has been deployed successfully and the **Status** will say ***Deployment succeeded***.

Go to **Platform Configuration > Conferencing Nodes** to return to the list of Conferencing Nodes.

After deploying a new Conferencing Node, it takes approximately 5 minutes before the node is available for conference hosting and for its status to be updated on the Management Node. (Until it is available, the Management Node will report the status of the Conferencing Node as having a last contacted and last updated date of "Never".)

Enabling automatic startup

After deploying a new Conferencing Node from VMware, you must enable automatic startup of that virtual machine (VM). In VMware, automatic startup is disabled by default for every new VM — which means that if the host server is powered down for any reason, when it restarts the VM will not restart and must be started manually.

You can only enable automatic startup after the Conferencing Node has been deployed.

Using the web client

To enable automatic startup using the vSphere web client:

1. Log in to the VM manager (vCenter Server).
2. From the left-hand panel, select **VCenter > Hosts And Clusters** and go to the host server on which the node's VM is installed.
3. Select the **Manage** tab.
4. From the pane on the left-hand side, select **VM Startup/Shutdown**.
5. At the top right of the page, select **Edit**.
6. Select the relevant node's VM and use the up arrow to move it to the **Automatic Startup** section.
7. Select **OK**.

Using the desktop client

To enable automatic startup using the vSphere desktop client:

1. Log in to the VM manager (vCenter Server or the ESXi host).
2. From the left-hand panel, select the host server on which the node's VM is installed.
3. Select the **Configuration** tab.
4. From the **Software** pane on the left-hand side, select **Virtual Machine Startup/Shutdown**.
5. At the top right of the page, select **Properties**.
6. Select the relevant VM and use the **Move Up** button to move it to the **Automatic Startup** section.
7. Select **OK**.

Disabling EVC

We strongly recommend that you disable EVC (Enhanced vMotion Compatibility) for any ESXi clusters hosting Conferencing Nodes that include a mix of old and new CPUs. If EVC is enabled on such clusters, the Pexip Infinity platform will run more slowly because the Conferencing Nodes will assume they are running on older hardware.

For more information, see Enhanced vMotion Compatibility (EVC).

To disable EVC:

1. Log in to vCenter Server using either the vSphere desktop or web client.
2. Display the cluster in the inventory.
3. Right-click the cluster and select **Edit Settings**.
4. In the left panel, select **VMware EVC**.
   The dialog box displays the current EVC settings.

5. Select **Change EVC Settings**.
6. Select **Disable EVC**.

Next steps

See Next steps after initial installation.

Manually deploying a Conferencing Node on an ESXi host

The manual deployment process generates an **.ova** file that must then be manually deployed from within VMware on to an ESXi host. Note that:

- This file is specific to the Conferencing Node being deployed. It cannot be used to deploy multiple Conferencing Nodes.
- The file is single-use. It cannot be used to re-deploy the same Conferencing Node at a later date. To re-deploy the Conferencing Node, you must first delete it from the Pexip Infinity Management Node and from VMware, and then deploy a new Conferencing Node with the same configuration as the deleted node.
- Before you start, ensure that you are currently using the same machine that you will subsequently want to use to upload the generated file on to your host server.

To manually deploy a new Conferencing Node on to a VMware ESXi host:

1. Go to **Platform Configuration > Conferencing Nodes** and select **Add Conferencing Node**.
2. From the **Deployment type** field, select either ***Manual (ESXi 6.0)***, ***Manual (ESXi 5.x)*** or ***Manual (ESXi 4.1)*** as appropriate.
3. Select **Next**.
4. You are now asked to provide information regarding the CPUs and memory of the Conferencing Node, by completing the following fields:

Option	Description
Number of virtual CPUs to assign	Enter the number of virtual CPUs to assign to the Conferencing Node. We recommend no more than one virtual CPU per physical core, unless you are making use of CPUs that support hyperthreading.
System memory (in megabytes) to assign	Enter the amount of RAM (in megabytes) to assign to the Conferencing Node. The number entered must be a multiple of 4. We recommend 1024 MB (1 GB) RAM for each virtual CPU.

5. Select **Next**.
6. You are now asked to provide the network configuration to be applied to the Conferencing Node, by completing the following fields:

Option	Description
Name	Enter the name that will be used to refer to this Conferencing Node in the Pexip Infinity Administrator interface.
Description	An optional field where you can provide more information about the Conferencing Node.
Hostname Domain	Enter the hostname and domain to be assigned to this Conferencing Node. Each Conferencing Node and Management Node must have a unique hostname. The Hostname and Domain together make up the Conferencing Node's DNS name or FQDN. We recommend you assign valid DNS names to all your Conferencing Nodes. For more information, see Assigning hostnames and FQDNs.
IPv4 address	Enter the IP address to be assigned to this Conferencing Node when it is created.
Network mask	Enter the IP network mask to be assigned to this Conferencing Node.
Gateway IPv4 address	Enter the IP address of the default gateway to be assigned to this Conferencing Node.

Option	Description
Secondary interface IPv4 address	The optional secondary interface IPv4 address for this Conferencing Node. If configured, this interface is used for signaling and media communications to clients, and the primary interface is used for communication with the Management Node and other Conferencing Nodes. For more information, see Conferencing Nodes with dual network interfaces (NICs).
Secondary interface network mask	The optional secondary interface network mask for this Conferencing Node.
System location	Select the physical location of this Conferencing Node. If the system location does not already exist, you can create a new one here by clicking ✚ to the right of the field. This will open up a new window showing the **Add System Location** page. For further information see About system locations.
SIP TLS FQDN	A unique identity for this Conferencing Node, used in signaling SIP TLS Contact addresses. For more information, see SIP TLS FQDN.
TLS certificate	The TLS certificate to use on this node. This must be a certificate that contains the above **SIP TLS FQDN**. Each certificate is shown in the format **<subject name> (<issuer>)**.
IPv6 address	The IPv6 address for this Conferencing Node. Each Conferencing Node must have a unique IPv6 address.
Gateway IPv6 address	The IPv6 address of the default gateway.
IPv4 static NAT address	The public IPv4 address used by this Conferencing Node when it is located behind a NAT device. Note that if you are using NAT, you must also configure your NAT device to route the Conferencing Node's **IPv4 static NAT address** to its **IPv4 address**. For more information, see Configuring Pexip Infinity nodes to work behind a static NAT device.
Static routes	From the list of **Available Static routes**, select the routes to assign to the node, and then use the right arrow to move the selected routes into the **Chosen Static routes** list. For more information, see Managing static routes.
Enable distributed database	This should usually be enabled (checked) for all Conferencing Nodes that are expected to be "always on", and disabled (unchecked) for nodes that are expected to only be powered on some of the time (e.g. nodes that are likely to only be operational during peak times).
SSH password	Enter the password to be used when logging in to this Conferencing Node's Linux operating system over SSH. The username will always be ***admin***. Logging in to the operating system is required when changing passwords or for diagnostic purposes only, and should generally be done under the guidance of your Pexip authorized support representative. In particular, do not change any configuration using SSH — all changes should be made using the Pexip Infinity Administrator interface.

7. Select **Finish**.
   You will be taken to the **Manually Deploy Conferencing Node** page.
8. Select **Download Conferencing Node**.
   A file with the name **pexip-<hostname>.<domain>.ova** will be downloaded.
9. When you wish to deploy the Conferencing Node VM, use a vSphere client to log in to vCenter Server (or the ESXi host directly, if it is not managed in vCenter Server) and select **File > Deploy OVF Template...**. Follow the on-screen prompts to deploy the **.ova** file.
   After deploying a new Conferencing Node, it takes approximately 5 minutes before the node is available for conference hosting and for its status to be updated on the Management Node. (Until it is available, the Management Node will report the status of the Conferencing Node as having a last contacted and last updated date of "Never".)

Enabling automatic startup

After deploying a new Conferencing Node from VMware, you must enable automatic startup of that virtual machine (VM). In VMware, automatic startup is disabled by default for every new VM — which means that if the host server is powered down for

any reason, when it restarts the VM will not restart and must be started manually.

You can only enable automatic startup after the Conferencing Node has been deployed.

Using the web client

To enable automatic startup using the vSphere web client:

a. Log in to the VM manager (vCenter Server).
b. From the left-hand panel, select **VCenter > Hosts And Clusters** and go to the host server on which the node's VM is installed.
c. Select the **Manage** tab.
d. From the pane on the left-hand side, select **VM Startup/Shutdown**.
e. At the top right of the page, select **Edit**.
f. Select the relevant node's VM and use the up arrow to move it to the **Automatic Startup** section.
g. Select **OK**.

Using the desktop client

To enable automatic startup using the vSphere desktop client:

a. Log in to the VM manager (vCenter Server or the ESXi host).
b. From the left-hand panel, select the host server on which the node's VM is installed.
c. Select the **Configuration** tab.
d. From the **Software** pane on the left-hand side, select **Virtual Machine Startup/Shutdown**.
e. At the top right of the page, select **Properties**.
f. Select the relevant VM and use the **Move Up** button to move it to the **Automatic Startup** section.
g. Select **OK**.

Disabling EVC

We strongly recommend that you disable EVC (Enhanced vMotion Compatibility) for any ESXi clusters hosting Conferencing Nodes that include a mix of old and new CPUs. If EVC is enabled on such clusters, the Pexip Infinity platform will run more slowly because the Conferencing Nodes will assume they are running on older hardware.

For more information, see Enhanced vMotion Compatibility (EVC).

To disable EVC:

1. Log in to vCenter Server using either the vSphere desktop or web client.
2. Display the cluster in the inventory.
3. Right-click the cluster and select **Edit Settings**.
4. In the left panel, select **VMware EVC**.
   The dialog box displays the current EVC settings.
5. Select **Change EVC Settings**.
6. Select **Disable EVC**.

Next steps

See Next steps after initial installation.

Manually deploying a Conferencing Node on a Hyper-V host

Before you start, ensure that you are currently using the same machine that you will subsequently want to use to upload the generated file on to your host server.

To manually deploy a new Conferencing Node on to a Hyper-V host:

1. Go to **Platform Configuration > Conferencing Nodes** and select **Add Conferencing Node**.
2. From the **Deployment type** field, select ***Manual (Hyper-V)***.

3. Select **Next**.
4. You are now asked to provide information regarding the CPUs and memory of the Conferencing Node VM, by completing the following fields:

Option	Description
Number of virtual CPUs to assign	Enter the number of virtual CPUs to assign to the Conferencing Node. We recommend no more than one virtual CPU per physical core, unless you are making use of CPUs that support hyperthreading.
System memory (in megabytes) to assign	Enter the amount of RAM (in megabytes) to assign to the Conferencing Node. The number entered must be a multiple of 4. We recommend 1024 MB (1 GB) RAM for each virtual CPU.

5. Select **Next**.
6. You are now asked to provide the network configuration to be applied to the Conferencing Node, by completing the following fields:

Option	Description
Name	Enter the name that will be used to refer to this Conferencing Node in the Pexip Infinity Administrator interface.
Description	An optional field where you can provide more information about the Conferencing Node.
Hostname Domain	Enter the hostname and domain to be assigned to this Conferencing Node. Each Conferencing Node and Management Node must have a unique hostname. The Hostname and Domain together make up the Conferencing Node's DNS name or FQDN. We recommend you assign valid DNS names to all your Conferencing Nodes. For more information, see Assigning hostnames and FQDNs.
IPv4 address	Enter the IP address to be assigned to this Conferencing Node when it is created.
Network mask	Enter the IP network mask to be assigned to this Conferencing Node.
Gateway IPv4 address	Enter the IP address of the default gateway to be assigned to this Conferencing Node.
Secondary interface IPv4 address	The optional secondary interface IPv4 address for this Conferencing Node. If configured, this interface is used for signaling and media communications to clients, and the primary interface is used for communication with the Management Node and other Conferencing Nodes. For more information, see Conferencing Nodes with dual network interfaces (NICs).
Secondary interface network mask	The optional secondary interface network mask for this Conferencing Node.
System location	Select the physical location of this Conferencing Node. If the system location does not already exist, you can create a new one here by clicking ✚ to the right of the field. This will open up a new window showing the **Add System Location** page. For further information see About system locations.
SIP TLS FQDN	A unique identity for this Conferencing Node, used in signaling SIP TLS Contact addresses. For more information, see SIP TLS FQDN.
TLS certificate	The TLS certificate to use on this node. This must be a certificate that contains the above **SIP TLS FQDN**. Each certificate is shown in the format **<subject name> (<issuer>)**.
IPv6 address	The IPv6 address for this Conferencing Node. Each Conferencing Node must have a unique IPv6 address.
Gateway IPv6 address	The IPv6 address of the default gateway.

Option	Description
IPv4 static NAT address	The public IPv4 address used by this Conferencing Node when it is located behind a NAT device. Note that if you are using NAT, you must also configure your NAT device to route the Conferencing Node's **IPv4 static NAT address** to its **IPv4 address**. For more information, see Configuring Pexip Infinity nodes to work behind a static NAT device.
Static routes	From the list of **Available Static routes**, select the routes to assign to the node, and then use the right arrow to move the selected routes into the **Chosen Static routes** list. For more information, see Managing static routes.
Enable distributed database	This should usually be enabled (checked) for all Conferencing Nodes that are expected to be "always on", and disabled (unchecked) for nodes that are expected to only be powered on some of the time (e.g. nodes that are likely to only be operational during peak times).
SSH password	Enter the password to be used when logging in to this Conferencing Node's Linux operating system over SSH. The username will always be ***admin***. Logging in to the operating system is required when changing passwords or for diagnostic purposes only, and should generally be done under the guidance of your Pexip authorized support representative. In particular, do not change any configuration using SSH — all changes should be made using the Pexip Infinity Administrator interface.

7. Select **Finish**.
   You will be taken to the **Manually Deploy Conferencing Node** page.
8. Select **Download Conferencing Node**.
   A zip file with the name **pexip-<hostname>.<domain>.zip** will be downloaded.
9. When you wish to deploy the Conferencing Nodes:
   a. Copy the zip file to the server running Hyper-V and unzip it.
      You will see a subfolder called **Virtual Machines**, containing an XML file which contains the configuration for the Conferencing Node VM.
   b. Open the Hyper-V Manager and select **Import Virtual Machine...**.
   c. Follow the on-screen prompts to deploy the Conferencing Node VM.
      When prompted, select the **Virtual Machines** folder and the Hyper-V manager will automatically discover the XML file.
      Select the type of import most appropriate for your environment (if you are unsure, select **Restore the virtual machine**).

After deploying a new Conferencing Node, it takes approximately 5 minutes before the node is available for conference hosting and for its status to be updated on the Management Node. (Until it is available, the Management Node will report the status of the Conferencing Node as having a last contacted and last updated date of "Never".)

To look at the steps taken in setting up an example lab deployment of a Conferencing Node in Microsoft Hyper-V, see http://www.graham-walsh.com/2016/02/setting-up-pexip-infinity-with-microsoft-hyper-v-server-2012/.

Enabling automatic startup

By default, virtual machines deployed using Hyper-V are configured to restart automatically if they were running when the host server was shut down or powered off. We recommend that you leave this setting as is for the Management Node and all Conferencing Nodes.

Disabling processor compatibility mode

We strongly recommend that you disable processor compatibility mode on all Hyper-V Conferencing Node virtual machines. If processor compatibility mode is enabled, the Conferencing Node may assume it is running on older hardware, and may stop working, with the message **CPU instruction set is not supported; system will be placed in maintenance mode**.

For more information, see **https://technet.microsoft.com/en-us/library/dn859550.aspx**.

Next steps

See Next steps after initial installation.

Manually deploying a Conferencing Node on a KVM host

To manually deploy a new Conferencing Node onto a KVM host, you must:

1. Use the Pexip Infinity Administrator interface to generate and download the .ova image.
2. Convert the .ova image for use with KVM.
3. Create a new volume on your KVM server and upload the disk image.
4. Create the Conferencing Node virtual machine.

   Note that we use the libvirt command line tools to perform the import as they provide greater control than Virtual Machine Manager.
5. Enable the virtual machine for automatic startup.

These steps are described in detail below.

Generate and download the .ova image

Before you start, ensure that you are currently using the same machine that you will subsequently want to use to upload the generated file on to your host server.

Use the Pexip Infinity Administrator interface to generate and download the .ova image:

1. Go to **Platform Configuration > Conferencing Nodes** and select **Add Conferencing Node**.
2. From the **Deployment type** field, select ***Manual (KVM)***.
3. Select **Next**.
4. You are now asked to provide information regarding the CPUs and memory of the Conferencing Node VM, by completing the following fields:

Option	Description
Number of virtual CPUs to assign	Enter the number of virtual CPUs to assign to the Conferencing Node. We recommend no more than one virtual CPU per physical core, unless you are making use of CPUs that support hyperthreading.
System memory (in megabytes) to assign	Enter the amount of RAM (in megabytes) to assign to the Conferencing Node. The number entered must be a multiple of 4. We recommend 1024 MB (1 GB) RAM for each virtual CPU.

5. Select **Next**.
6. You are now asked to provide the network configuration to be applied to the Conferencing Node, by completing the following fields:

Option	Description
Name	Enter the name that will be used to refer to this Conferencing Node in the Pexip Infinity Administrator interface.
Description	An optional field where you can provide more information about the Conferencing Node.
Hostname Domain	Enter the hostname and domain to be assigned to this Conferencing Node. Each Conferencing Node and Management Node must have a unique hostname. The Hostname and Domain together make up the Conferencing Node's DNS name or FQDN. We recommend you assign valid DNS names to all your Conferencing Nodes. For more information, see Assigning hostnames and FQDNs.
IPv4 address	Enter the IP address to be assigned to this Conferencing Node when it is created.
Network mask	Enter the IP network mask to be assigned to this Conferencing Node.
Gateway IPv4 address	Enter the IP address of the default gateway to be assigned to this Conferencing Node.

Option	Description
Secondary interface IPv4 address	The optional secondary interface IPv4 address for this Conferencing Node. If configured, this interface is used for signaling and media communications to clients, and the primary interface is used for communication with the Management Node and other Conferencing Nodes. For more information, see Conferencing Nodes with dual network interfaces (NICs).
Secondary interface network mask	The optional secondary interface network mask for this Conferencing Node.
System location	Select the physical location of this Conferencing Node. If the system location does not already exist, you can create a new one here by clicking ✚ to the right of the field. This will open up a new window showing the **Add System Location** page. For further information see About system locations.
SIP TLS FQDN	A unique identity for this Conferencing Node, used in signaling SIP TLS Contact addresses. For more information, see SIP TLS FQDN.
TLS certificate	The TLS certificate to use on this node. This must be a certificate that contains the above **SIP TLS FQDN**. Each certificate is shown in the format **<subject name> (<issuer>)**.
IPv6 address	The IPv6 address for this Conferencing Node. Each Conferencing Node must have a unique IPv6 address.
Gateway IPv6 address	The IPv6 address of the default gateway.
IPv4 static NAT address	The public IPv4 address used by this Conferencing Node when it is located behind a NAT device. Note that if you are using NAT, you must also configure your NAT device to route the Conferencing Node's **IPv4 static NAT address** to its **IPv4 address**. For more information, see Configuring Pexip Infinity nodes to work behind a static NAT device.
Static routes	From the list of **Available Static routes**, select the routes to assign to the node, and then use the right arrow to move the selected routes into the **Chosen Static routes** list. For more information, see Managing static routes.
Enable distributed database	This should usually be enabled (checked) for all Conferencing Nodes that are expected to be "always on", and disabled (unchecked) for nodes that are expected to only be powered on some of the time (e.g. nodes that are likely to only be operational during peak times).
SSH password	Enter the password to be used when logging in to this Conferencing Node's Linux operating system over SSH. The username will always be ***admin***. Logging in to the operating system is required when changing passwords or for diagnostic purposes only, and should generally be done under the guidance of your Pexip authorized support representative. In particular, do not change any configuration using SSH — all changes should be made using the Pexip Infinity Administrator interface.

7. Select **Finish**.
   You are taken to the **Manually Deploy Conferencing Node** page.
8. Select **Download Conferencing Node**.
   This downloads a zip file with the name **pexip-<hostname>.<domain>.ova**.

Convert the .ova image for use with KVM

To use the Conferencing Node OVA image file with KVM, you must convert it from VMDK to raw format:

1. Copy the downloaded OVA file (named **pexip-<hostname>.<domain>.ova**) to the server running KVM.
2. Unpack the .ova image, using the command:
   ```
   tar xf pexip-<hostname>.<domain>.ova
   ```
   This unpacks a set of files including **pexip-disk01.vmdk**.
3. Convert the disk image from VMDK to raw, using the command:

```
qemu-img convert -O raw pexip-disk01.vmdk pexip-disk01.raw
```

(This conversion process can take several seconds.)

Create a new volume and upload the disk image

Next, you create a new volume on your KVM server and upload the converted disk image. From within your KVM environment:

1. Use **virsh** to create a new volume on your KVM server:
   ```
   virsh --connect qemu://<hostname>/system vol-create-as <poolname> <volume_name> 49G --format raw
   ```
   where:
   `<hostname>` is the hostname of your KVM server. Note that you can omit the `<hostname>` if you are running **virsh** commands on the local server i.e. you can use `virsh --connect qemu:///system`.
   `<poolname>` is the name of the storage pool in which to create the volume; typically you would use **default**. (To determine the storage pools available on the target system, use `virsh --connect qemu://<hostname>/system pool-list`.)
   `<volume_name>` is the name of your new volume.
   `49G` is the virtual size of the volume; always use 49G for a Conferencing Node.
   For example:
   ```
   virsh --connect qemu://host1.example.com/system vol-create-as default pexip-conf-01 49G --format raw
   ```
   This example creates a volume named **pexip-conf-01** of size 49 GB and format raw in the storage pool named **default**.
2. Upload the converted disk image to the newly created volume:
   ```
   virsh --connect qemu://<hostname>/system vol-upload <volume_name> pexip-disk01.raw --pool <poolname>
   ```
   For example:
   ```
   virsh --connect qemu://host1.example.com/system vol-upload pexip-conf-01 pexip-disk01.raw --pool default
   ```
   This example uploads the **pexip-disk01.raw** image to the newly created volume, **pexip-conf-01**, in the storage pool named **default**.

Create the virtual machine

After the disk image has been uploaded, you can create the virtual machine to use it.

Note that we use the libvirt command line tools to perform the import as they provide greater control than Virtual Machine Manager.

1. Identify the filesystem path of the newly uploaded disk image:
   ```
   virsh --connect qemu://<hostname>/system vol-path <volume_name> --pool <poolname>
   ```
   For example:
   ```
   virsh --connect qemu://host1.example.com/system vol-path pexip-conf-01 --pool default
   ```
   This prints out the absolute path to the disk image file, for example:
   ```
   /var/lib/libvirt/images/pexip-conf-01
   ```
   This path is used in the **disk path** parameter in the next step.
2. Use the **virt-install** command line tool to create the virtual machine:
   ```
   virt-install \
     --import \
     --hvm \
     --name=<vm_name> \
     --arch=x86_64 \
     --vcpus=4 \
     --ram=4096 \
     --cpu host \
     --os-type=linux \
     --connect=qemu://<hostname>/system \
     --virt-type kvm \
     --disk path=<image_file_path>,bus=virtio,format=raw,cache=none,io=native \
     --network bridge=br0,model=virtio \
   ```

```
  --memballoon virtio \
  --graphics vnc,listen=0.0.0.0,password=<password>
```

This creates a new VM (KVM domain) from the converted disk image.

The meaning of the various options are as follows (items in **bold** may be changed as necessary):

Option	Description
--import	Build guest domain around pre-installed disk image; do not attempt to install a new OS.
--hvm	Create a fully virtualized (i.e. not paravirtualized) VM.
--name=**<vm_name>**	Name of the new VM, where <vm_name> is, for example, **pexip-conf01-vm**.
--arch=x86_64	CPU architecture of new VM (must be x84_64).
--vcpus=**4**	Number of CPUs allocated to new VM. By default, this is **4** for the Conferencing Node.
--ram=**4096**	Memory allocated to new VM (in megabytes).
--cpu host	Expose all host CPU capabilities to new VM (CPUID).
--os-type=linux	The guest OS is Linux.
--connect=qemu://**<hostname>**/system	Connect to KVM on the target system, where <hostname> is the hostname of your KVM server.
--virt-type kvm	Use KVM to host the new VM.
--disk path=**<image_file_path>**,bus=virtio,format=raw,cache=none,io=native	○ Define the location of the disk image file, where <image_file_path> is as determined in the previous step, for example **/var/lib/libvirt/images/pexip-conf-01**. ○ Expose it to the guest on the virtio paravirtualized bus (as opposed to IDE/SCSI). ○ Define the image file as being in raw format. ○ Instruct the host system not to cache the disk contents in memory. ○ Use the native IO backend to access the disk device.
--network **bridge=br0,model=virtio**	○ Create a network interface connected to the **br0** bridge interface on the host. ○ Expose it to the guest as a **virtio** paravirtualized NIC.
--memballoon virtio	Expose the virtio memory balloon to the guest.
--graphics vnc,listen=**0.0.0.0**,password=**<password>**	Expose the graphical console over VNC, listening on **0.0.0.0** (i.e. all addresses on the target system) and with an access password of **<password>**.

You may receive a warning "Unable to connect to graphical console: virt-viewer not installed"; if so, this message can be safely ignored.

After the VM has been created, it may be managed using the Virtual Machine Manager desktop interface (virt-manager application) or via the command line interface (virsh).

The new node should start automatically. If it does not you can use the Virtual Machine Manager to start the node, or the CLI command:

```
virsh --connect qemu://<hostname>/system start <vm_name>
```

Note that you can list existing VMs by using `virsh --connect qemu://<hostname>/system list`

After deploying a new Conferencing Node, it takes approximately 5 minutes before the node is available for conference hosting and for its status to be updated on the Management Node. (Until it is available, the Management Node will report the status of the Conferencing Node as having a last contacted and last updated date of "Never".)

Enabling automatic startup

After deploying a new Conferencing Node in KVM, you should enable automatic startup of that virtual machine (VM). In KVM, automatic startup is disabled by default for every new VM. This means that if the host server is powered down for any reason, when it restarts the VM will not restart and must be started manually.

You can only enable automatic startup after the Conferencing Node has been deployed.

To enable automatic startup using Virtual Machine Manager:

1. Connect to the Virtual Machine Manager (virt-manager) that is managing the node's VM.
2. Select the node's VM and then, from the toolbar, select the **Show the virtual machine console and details** icon .
   A new window for that VM is opened.
3. If necessary, select **View > Details** to display the VM information.
4. From the sidebar menu, select **Boot Options**.
5. Select the **Start virtual machine on host boot up** check box.
6. Select **Apply**.

Next steps

See Next steps after initial installation.

Manually deploying a Conferencing Node on a Xen host

To manually deploy a new Conferencing Node onto a Xen host, you must:

1. Use the Pexip Infinity Administrator interface to generate and download the .ova image.
2. Convert the .ova image for use with Xen.
3. Create a new volume on your Xen server and upload the disk image.
4. Create the Conferencing Node virtual machine.
   Note that we use the libvirt command line tools to perform the import as they provide greater control than Virtual Machine Manager.
5. Enable the virtual machine for automatic startup.

These steps are described in detail below.

Generate and download the .ova image

Before you start, ensure that you are currently using the same machine that you will subsequently want to use to upload the generated file on to your host server.

Use the Pexip Infinity Administrator interface to generate and download the .ova image:

1. Go to **Platform Configuration > Conferencing Nodes** and select **Add Conferencing Node**.
2. From the **Deployment type** field, select ***Manual (Xen)***.
3. Select **Next**.
4. You are now asked to provide information regarding the CPUs and memory of the Conferencing Node VM, by completing the following fields:

Option	Description
Number of virtual CPUs to assign	Enter the number of virtual CPUs to assign to the Conferencing Node. We recommend no more than one virtual CPU per physical core, unless you are making use of CPUs that support hyperthreading.
System memory (in megabytes) to assign	Enter the amount of RAM (in megabytes) to assign to the Conferencing Node. The number entered must be a multiple of 4. We recommend 1024 MB (1 GB) RAM for each virtual CPU.

5. Select **Next**.

6. You are now asked to provide the network configuration to be applied to the Conferencing Node, by completing the following fields:

Option	Description
Name	Enter the name that will be used to refer to this Conferencing Node in the Pexip Infinity Administrator interface.
Description	An optional field where you can provide more information about the Conferencing Node.
Hostname Domain	Enter the hostname and domain to be assigned to this Conferencing Node. Each Conferencing Node and Management Node must have a unique hostname. The Hostname and Domain together make up the Conferencing Node's DNS name or FQDN. We recommend you assign valid DNS names to all your Conferencing Nodes. For more information, see Assigning hostnames and FQDNs.
IPv4 address	Enter the IP address to be assigned to this Conferencing Node when it is created.
Network mask	Enter the IP network mask to be assigned to this Conferencing Node.
Gateway IPv4 address	Enter the IP address of the default gateway to be assigned to this Conferencing Node.
Secondary interface IPv4 address	The optional secondary interface IPv4 address for this Conferencing Node. If configured, this interface is used for signaling and media communications to clients, and the primary interface is used for communication with the Management Node and other Conferencing Nodes. For more information, see Conferencing Nodes with dual network interfaces (NICs).
Secondary interface network mask	The optional secondary interface network mask for this Conferencing Node.
System location	Select the physical location of this Conferencing Node. If the system location does not already exist, you can create a new one here by clicking ✚ to the right of the field. This will open up a new window showing the **Add System Location** page. For further information see About system locations.
SIP TLS FQDN	A unique identity for this Conferencing Node, used in signaling SIP TLS Contact addresses. For more information, see SIP TLS FQDN.
TLS certificate	The TLS certificate to use on this node. This must be a certificate that contains the above **SIP TLS FQDN**. Each certificate is shown in the format **<subject name> (<issuer>)**.
IPv6 address	The IPv6 address for this Conferencing Node. Each Conferencing Node must have a unique IPv6 address.
Gateway IPv6 address	The IPv6 address of the default gateway.
IPv4 static NAT address	The public IPv4 address used by this Conferencing Node when it is located behind a NAT device. Note that if you are using NAT, you must also configure your NAT device to route the Conferencing Node's **IPv4 static NAT address** to its **IPv4 address**. For more information, see Configuring Pexip Infinity nodes to work behind a static NAT device.
Static routes	From the list of **Available Static routes**, select the routes to assign to the node, and then use the right arrow to move the selected routes into the **Chosen Static routes** list. For more information, see Managing static routes.
Enable distributed database	This should usually be enabled (checked) for all Conferencing Nodes that are expected to be "always on", and disabled (unchecked) for nodes that are expected to only be powered on some of the time (e.g. nodes that are likely to only be operational during peak times).
SSH password	Enter the password to be used when logging in to this Conferencing Node's Linux operating system over SSH. The username will always be ***admin***. Logging in to the operating system is required when changing passwords or for diagnostic purposes only, and should generally be done under the guidance of your Pexip authorized support representative. In particular, do not change any configuration using SSH — all changes should be made using the Pexip Infinity Administrator interface.

7. Select **Finish**.
   You are taken to the **Manually Deploy Conferencing Node** page.
8. Select **Download Conferencing Node**.
   This downloads a zip file with the name **pexip-<hostname>.<domain>.ova**.

Convert the .ova image for use with Xen

To use the Conferencing Node OVA image file with Xen, you must convert it from VMDK to raw format:

1. Copy the downloaded OVA file (named **pexip-<hostname>.<domain>.ova**) to the server running Xen.
2. Unpack the .ova image, using the command:
   ```
   tar xf pexip-<hostname>.<domain>.ova
   ```
   This unpacks a set of files including **pexip-disk01.vmdk**.
3. Convert the disk image from VMDK to raw, using the command:
   ```
   qemu-img convert -O raw pexip-disk01.vmdk pexip-disk01.raw
   ```
   (This conversion process can take several seconds.)

Create a new volume and upload the disk image

Next, you create a new volume on your Xen server and upload the converted disk image. From within your Xen environment:

1. Use **virsh** to create a new volume on your Xen server:
   ```
   virsh --connect xen://<hostname>/ vol-create-as <poolname> <volume_name> 49G --format raw
   ```
   where:
   `<hostname>` is the hostname of your Xen server. Note that you can omit the `<hostname>` if you are running **virsh** commands on the local server i.e. you can use `virsh --connect xen:///`.
   `<poolname>` is the name of the storage pool in which to create the volume; typically you would use **default**. (To determine the storage pools available on the target system, use `virsh --connect xen://<hostname>/ pool-list`.)
   `<volume_name>` is the name of your new volume.
   `49G` is the virtual size of the volume; always use 49G for a Conferencing Node.
   For example:
   ```
   virsh --connect xen://host1.example.com/ vol-create-as default pexip-conf-01 49G --format raw
   ```
   This example creates a volume named **pexip-conf-01** of size 49 GB and format raw in the storage pool named **default**.
2. Upload the converted disk image to the newly created volume:
   ```
   virsh --connect xen://<hostname>/ vol-upload <volume_name> pexip-disk01.raw --pool <poolname>
   ```
   For example:
   ```
   virsh --connect xen://host1.example.com/ vol-upload pexip-conf-01 pexip-disk01.raw --pool default
   ```
   This example uploads the **pexip-disk01.raw** image to the newly created volume, **pexip-conf-01**, in the storage pool named **default**.

Create the virtual machine

After the disk image has been uploaded, you can create the virtual machine to use it.

Note that we use the libvirt command line tools to perform the import as they provide greater control than Virtual Machine Manager.

1. Identify the filesystem path of the newly uploaded disk image:
   ```
   virsh --connect xen://<hostname>/ vol-path <volume_name> --pool <poolname>
   ```
   For example:
   ```
   virsh --connect xen://host1.example.com/ vol-path pexip-conf-01 --pool default
   ```
   This prints out the absolute path to the disk image file, for example:
   ```
   /var/lib/libvirt/images/pexip-conf-01
   ```

This path is used in the **disk path** parameter in the next step.

2. Use the **virt-install** command line tool to create the virtual machine:

```
virt-install \
  --import \
  --hvm \
  --name=<vm_name> \
  --arch=x86_64 \
  --vcpus=4 \
  --ram=4096 \
  --os-type=linux \
  --connect=xen://<hostname>/ \
  --virt-type xen \
  --disk path=<image_file_path>,bus=xen,format=raw,driver_name=qemu,cache=none,io=native \
  --network bridge=xenbr0,model=e1000 \
  --memballoon xen \
  --graphics vnc,listen=0.0.0.0,password=<password>
```

This creates a new VM (Xen domain) from the converted disk image.

The meaning of the various options are as follows (items in **bold** may be changed as necessary):

Option	Description
--import	Build guest domain around pre-installed disk image; do not attempt to install a new OS.
--hvm	Create a fully virtualized (i.e. not paravirtualized) VM.
--name=**<vm_name>**	Name of the new VM, where <vm_name> is, for example, **pexip-conf01-vm**.
--arch=x86_64	CPU architecture of new VM (must be x84_64).
--vcpus=**4**	Number of CPUs allocated to new VM. By default, this is **4** for the Conferencing Node.
--ram=**4096**	Memory allocated to new VM (in megabytes).
--os-type=linux	The guest OS is Linux.
--connect=xen://**<hostname>**/	Connect to Xen on the target system, where <hostname> is the hostname of your Xen server.
--virt-type xen	Use Xen to host the new VM.
--disk path=**<image_file_path>**,bus=xen,format=raw,driver_name=qemu,cache=none,io=native	○ Define the location of the disk image file, where <image_file_path> is as determined in the previous step, for example **/var/lib/libvirt/images/pexip-conf-01**. ○ Expose it to the guest on the xen paravirtualized bus (as opposed to IDE/SCSI). ○ Define the image file as being in raw format and the driver as qemu. ○ Instruct the host system not to cache the disk contents in memory. ○ Use the native IO backend to access the disk device.
--network **bridge=xenbr0,model=e1000**	○ Create a network interface connected to the **xenbr0** bridge interface on the host. ○ Expose it to the guest as an **e1000** NIC.
--memballoon xen	Expose the xen memory balloon to the guest.
--graphics vnc,listen=**0.0.0.0**,password=**<password>**	Expose the graphical console over VNC, listening on **0.0.0.0** (i.e. all addresses on the target system) and with an access password of **<password>**.

You may receive a warning "Unable to connect to graphical console: virt-viewer not installed"; if so, this message can be safely ignored.

After the VM has been created, it may be managed using the Virtual Machine Manager desktop interface (virt-manager application) or via the command line interface (virsh).

The new node should start automatically. If it does not you can use the Virtual Machine Manager to start the node, or the CLI command:

```
virsh --connect xen://<hostname>/ start <vm_name>
```

Note that you can list existing VMs by using `virsh --connect xen://<hostname>/ list`

After deploying a new Conferencing Node, it takes approximately 5 minutes before the node is available for conference hosting and for its status to be updated on the Management Node. (Until it is available, the Management Node will report the status of the Conferencing Node as having a last contacted and last updated date of "Never".)

Enabling automatic startup

After deploying a new Conferencing Node in Xen, you should enable automatic startup of that virtual machine (VM). In Xen, automatic startup is disabled by default for every new VM. This means that if the host server is powered down for any reason, when it restarts the VM will not restart and must be started manually.

You can only enable automatic startup after the Conferencing Node has been deployed.

To enable automatic startup using Virtual Machine Manager:

1. Connect to the Virtual Machine Manager (virt-manager) that is managing the node's VM.
2. Select the node's VM and then, from the toolbar, select the **Show the virtual machine console and details** icon .
   A new window for that VM is opened.
3. If necessary, select **View > Details** to display the VM information.
4. From the sidebar menu, select **Boot Options**.
5. Select the **Start virtual machine on host boot up** check box.
6. Select **Apply**.

Next steps

See Next steps after initial installation.

Deploying a Conferencing Node using a generic VM template and configuration file

Pexip also provide the Conferencing Node VM template and the ability to generate a Conferencing Node-specific configuration file for you to deploy yourself using non-supported hypervisors or orchestration layers.

Creating a new generic Conferencing Node is a two-step process:

1. A generic instance of a Conferencing Node VM is created using a template provided by Pexip.
2. The VM is configured with the details of the specific Conferencing Node being deployed.

Prerequisites

- The Conferencing Node must be deployed in a VM environment that supports address assignment by DHCP.
- You must know the IP address that will initially be assigned to the Conferencing Node. You will use this IP address to connect to the VM in order to upload the configuration file, but this configuration file may then assign a new IP address to the Conferencing Node.

Deployment process

To create a new Conferencing Node using the VM template:

1. Within your chosen environment, go to https://dl.pexip.com/infinity/index.html, select the appropriate directory for your software version, and then download one of the following files and use it to create a generic instance of a Conferencing Node:
   - **Pexip_Infinity_v14.1_generic_ConfNode_<build>.ova** in environments that take .ova or .ovf files as input.
   - **Pexip_Infinity_v14.1_HyperV_ConfNode_<build>.zip** for Hyper-V in cloud-based environments or other orchestration layers where standard manual deployment is problematic.

Ensure you are using a VM template with the same Pexip Infinity software version as that which is currently running on the Management Node. This includes dot releases — so for example, for a v12.1 Management Node you must install a v12.1 Conferencing Node rather than a v12 Conferencing Node. If the Management Node has been upgraded, you will need to download the Conferencing Node VM template corresponding to that software version. For more information, see Upgrading configuration-only deployments.

2. Go to **Platform Configuration > Conferencing Nodes** and select **Add Conferencing Node**.
3. From the **Deployment type** field, select ***Generic (configuration only)***.
4. Select **Next**.
5. You are now asked to provide the network configuration to be applied to the Conferencing Node, by completing the following fields:

Option	Description
Name	Enter the name that will be used to refer to this Conferencing Node in the Pexip Infinity Administrator interface.
Description	An optional field where you can provide more information about the Conferencing Node.
Hostname Domain	Enter the hostname and domain to be assigned to this Conferencing Node. Each Conferencing Node and Management Node must have a unique hostname. The Hostname and Domain together make up the Conferencing Node's DNS name or FQDN. We recommend you assign valid DNS names to all your Conferencing Nodes. For more information, see Assigning hostnames and FQDNs.
IPv4 address	Enter the IP address to be assigned to this Conferencing Node when it is created.
Network mask	Enter the IP network mask to be assigned to this Conferencing Node.
Gateway IPv4 address	Enter the IP address of the default gateway to be assigned to this Conferencing Node.
Secondary interface IPv4 address	The optional secondary interface IPv4 address for this Conferencing Node. If configured, this interface is used for signaling and media communications to clients, and the primary interface is used for communication with the Management Node and other Conferencing Nodes. For more information, see Conferencing Nodes with dual network interfaces (NICs).
Secondary interface network mask	The optional secondary interface network mask for this Conferencing Node.
System location	Select the physical location of this Conferencing Node. If the system location does not already exist, you can create a new one here by clicking ✚ to the right of the field. This will open up a new window showing the **Add System Location** page. For further information see About system locations.
SIP TLS FQDN	A unique identity for this Conferencing Node, used in signaling SIP TLS Contact addresses. For more information, see SIP TLS FQDN.
TLS certificate	The TLS certificate to use on this node. This must be a certificate that contains the above **SIP TLS FQDN**. Each certificate is shown in the format **<subject name> (<issuer>)**.
IPv6 address	The IPv6 address for this Conferencing Node. Each Conferencing Node must have a unique IPv6 address.
Gateway IPv6 address	The IPv6 address of the default gateway.
IPv4 static NAT address	The public IPv4 address used by this Conferencing Node when it is located behind a NAT device. Note that if you are using NAT, you must also configure your NAT device to route the Conferencing Node's **IPv4 static NAT address** to its **IPv4 address**. For more information, see Configuring Pexip Infinity nodes to work behind a static NAT device.

Option	Description
Static routes	From the list of **Available Static routes**, select the routes to assign to the node, and then use the right arrow to move the selected routes into the **Chosen Static routes** list. For more information, see Managing static routes.
Enable distributed database	This should usually be enabled (checked) for all Conferencing Nodes that are expected to be "always on", and disabled (unchecked) for nodes that are expected to only be powered on some of the time (e.g. nodes that are likely to only be operational during peak times).
SSH password	Enter the password to be used when logging in to this Conferencing Node's Linux operating system over SSH. The username will always be ***admin***. Logging in to the operating system is required when changing passwords or for diagnostic purposes only, and should generally be done under the guidance of your Pexip authorized support representative. In particular, do not change any configuration using SSH — all changes should be made using the Pexip Infinity Administrator interface.

6. Select **Finish**.
   You will be taken to the **Manually Deploy Conferencing Node** page.
7. Select **Download Conferencing Node Configuration**.
   A zip file with the name **pexip-<hostname>.<domain>.xml** will be downloaded.
8. Browse to **https://<conferencing-node-ip>:8443/** and use the form provided to upload the configuration file to the Conferencing Node VM.
   The Conferencing Node will apply the configuration and then reboot. When it has rebooted, it will connect to the Management Node in the usual way.
9. Within your hypervisor, configure the Conferencing Node VM with information regarding the CPUs and memory to be assigned to it, as follows:

Option	Description
Number of virtual CPUs to assign	Enter the number of virtual CPUs to assign to the Conferencing Node. We recommend no more than one virtual CPU per physical core, unless you are making use of CPUs that support hyperthreading.
System memory (in megabytes) to assign	Enter the amount of RAM (in megabytes) to assign to the Conferencing Node. The number entered must be a multiple of 4. We recommend 1024 MB (1 GB) RAM for each virtual CPU.

After deploying a new Conferencing Node, it takes approximately 5 minutes before the node is available for conference hosting and for its status to be updated on the Management Node. (Until it is available, the Management Node will report the status of the Conferencing Node as having a last contacted and last updated date of "Never".)

Enabling automatic startup

We recommend that all virtual machines are configured to restart automatically if they were running when the host server was shut down or powered off.

Next steps

See Next steps after initial installation.

Assigning hostnames and FQDNs

The Management Node and each Conferencing Node within your deployment must have a unique hostname.

Hostnames can be made of a combination of numbers, letters, and the hyphen (-) character, and are case-insensitive. Hostnames cannot be entirely numeric.

The hostname and the domain together make up the system's DNS name or FQDN.

We recommend that you assign valid DNS names or FQDNs to all of your Conferencing Nodes. Doing so will make managing your deployment easier, for example by:

- allowing you to route calls more easily
- allowing you to use more manageable TLS certificates
- allowing you to in effect change the IP address of a Conferencing Node (by re-deploying a new system using the same DNS name but a different IP address)
- making the system easier to access for the purposes of troubleshooting.

To assign a DNS name to a Conferencing Node, enter a valid **Hostname** and **Domain** combination when first deploying the Conferencing Node.

Configuring existing Conferencing Nodes

To reconfigure or delete an existing Conferencing Node, go to **Platform Configuration > Conferencing Nodes** and click on the name of the Conferencing Node.

Any changes to the configuration of the Conferencing Node should be made using the Pexip Infinity Administrator interface. Do not make any changes using other tools (such as VMware or Hyper-V); doing so may cause the Conferencing Node to fail. The only exception is any change to the CPU and RAM allocated to the Conferencing Node — these changes should be made using the hypervisor and should only be done while the Conferencing Node is powered off.

When reconfiguring existing Conferencing Nodes, the following information can be changed:

Option	Description
Name	The name used to refer to this Conferencing Node in the Pexip Infinity Administrator interface. Each Conferencing Node must have a unique name.
Description	An optional field where you can provide more information about the Conferencing Node.
System location	The physical location of this Conferencing Node. If you change the system location of an existing Conferencing Node, all existing calls will be disconnected and the Conferencing Node will be restarted.
Enable maintenance mode	While maintenance mode is enabled, this Conferencing Node will not accept any new conference instances. For more information, see About maintenance mode. The maintenance mode setting will persist after a reboot.
SIP TLS FQDN	A unique identity for this Conferencing Node, used in signaling SIP TLS Contact addresses. For more information, see SIP TLS FQDN.
TLS certificate	The TLS certificate to use on this node. This must be a certificate that contains the above **SIP TLS FQDN**. Each certificate is shown in the format **<subject name> (<issuer>)**.
IPv6 address	The IPv6 address for this Conferencing Node.
Gateway IPv6 address	The IPv6 address of the default gateway.
IPv4 static NAT address	The public IPv4 address used by this Conferencing Node when it is located behind a NAT device.
Static routes	From the list of **Available Static routes**, select the routes to assign to the node, and then use the right arrow to move the selected routes into the **Chosen Static routes** list. For more information, see Managing static routes.

You can also change the node's SNMP settings (see Enabling SNMP for more information):

Option	Description
SNMP mode	Configures the SNMP mode for the selected node: ***Off***: SNMP is disabled. You will not be able to use SNMP to query the node for its status. ***SNMPv2c read-only***: enables insecure, read-only access. ***SNMPv3 read-only***: enables secure, read-only access, using the **authPriv** security level with SHA1 authentication and AES 128-bit encryption. When enabled, access is provided to the basic RFC 1213 MIB-II tree (1.3.6.1.2.1). Default: ***Off***.
SNMP community	The SNMP group to which this node belongs. This setting applies to SNMPv2c only. Default: public
SNMPv3 username	The node's SNMPv3 username, used to authenticate SNMPv3 requests.
SNMPv3 privacy password	The node's SNMPv3 privacy password used for encrypting messages between the node and the management station. AES encryption must be used; DES is not supported.
SNMPv3 authentication password	The node's SNMPv3 authentication password, used to authenticate the associated username. The SHA authentication protocol must be used; MD5 is not supported.
SNMP system contact	The contact details (for example, email address) of the person responsible for this particular node.
SNMP system location	A description of the node's location.

Other details of the Conferencing Node that cannot be directly changed (however, see Changing the node's IP address below) are also shown on this page for your information, as follows:

Option	Description
IPv4 address	The IPv4 address of this Conferencing Node.
Network mask	The IP network mask for this Conferencing Node.
Gateway IPv4 address	The IPv4 address of the default gateway.
Secondary interface IPv4 address and network mask	The optional secondary interface IPv4 address and network mask for this Conferencing Node. These fields are only displayed if the Conferencing Node has been deployed with dual network interfaces.
Hostname Domain	The DNS hostname and domain of this Conferencing Node. Together these make up the machine's FQDN, or ***DNS Name*** in VMware. For more information, see Assigning hostnames and FQDNs.
Enable distributed database	This should usually be enabled (checked) for all Conferencing Nodes that are expected to be "always on", and disabled (unchecked) for nodes that are expected to only be powered on some of the time (e.g. nodes that are likely to only be operational during peak times). After a node has been deployed, this setting can only be changed via the Management configuration API using the worker_vm resource.

SIP TLS FQDN

If your deployment includes Microsoft Lync / Skype for Business, you **must** configure each Conferencing Node with a **SIP TLS FQDN**. However, for security purposes we recommend that all deployments use SIP TLS FQDNs.

To configure the **SIP TLS FQDN**, go to **Platform Configuration > Conferencing Nodes** and select each Conferencing Node in turn.

The FQDN in the **SIP TLS FQDN** field is used in SIP signaling over TLS, and specifies the identity that the Conferencing Node service will use when identifying itself to the systems connecting to it. Each Conferencing Node must have a unique **SIP TLS FQDN**.

The **SIP TLS FQDN**:

- must match one of the identities returned in the certificate for the Conferencing Node, and
- must have an entry in DNS

so that the identity of the Conferencing Node can be verified by the systems connecting to it.

The **SIP TLS FQDN** can be the same as the Conferencing Node's FQDN (made up of its Hostname and Domain).

If the **SIP TLS FQDN** field is left blank, the IP address of the Conferencing Node will be used in SIP TLS signaling, and, depending on your call control configuration, this may result in calls failing.

For more details on the use of domain certificates in SIP, see **section 4 of RFC 5922**.

Changing the node's IP address

You can change the IPv4 address of a Conferencing Node by adding a new node with the new address, and deleting the node with the old address.

If you want to preserve the Conferencing Node's existing FQDN, you should:

1. Put the Conferencing Node whose IP address you want to change into maintenance mode.
2. When all calls on that Conferencing Node have terminated, delete the Conferencing Node.
3. Wait at least 60 seconds to allow the deletion to be synchronized across the platform.
4. Add a new Conferencing Node with the new IP address (but using the same **Name**, **Hostname** and **Domain** etc. as used before, if required).
5. If necessary, make any changes to your call control system so that calls are routed to the Conferencing Node's new IP address.

If you want to maintain full service capacity, and do not need to preserve the node's FQDN, then you can add the new node with the new IP address before deleting the old node with the old IP address. There is no limit on the number of Conferencing Nodes that you can add to the Pexip Infinity platform.

Do not attempt to change the IP address of a Conferencing Node using utilities available in external tools (such as VMware or Hyper-V) because this will cause Pexip Infinity services to fail.

Deleting Conferencing Nodes

When deleting a Conferencing Node, you must first remove its details from the Pexip Infinity platform using the Pexip Infinity Administrator interface, and then delete it from the hypervisor/cloud platform.

To delete a Conferencing Node:

1. Put the Conferencing Node into maintenance mode and wait until all calls on it have terminated.
2. From the Pexip Infinity Administrator interface, go to **Platform Configuration > Conferencing Nodes**.
3. Select the Conferencing Node to be deleted, and from the **Action** drop-down menu select ***Delete selected Conferencing Nodes***.
4. Select **Go** and on the following page confirm that you want to delete the selected Conferencing Node by selecting **Yes, I'm sure**.
5. Shut down and remove the Conferencing Node VM.
   For on-premises installations:
   a. Log in to the VM manager, shut down the deleted Conferencing Node and then power it off.
   b. Right-click on the Conferencing Node and select ***Delete from Disk*** (VMware) or ***Delete*** (Hyper-V / KVM / Xen).
   For cloud-based installations:
   a. Log in to your cloud provider's management portal.
   b. Terminate the Conferencing Node instance.
   c. If using Azure, delete the resource group and storage containers which hold the instance.

Pexip Infinity services

About Pexip Infinity services

The **Service Configuration** menu allows you to manage the conference services offered by Pexip Infinity. These services include:

- Virtual Meeting Rooms: personal meeting spaces in which you can hold conferences.
- Virtual Auditoriums: larger meeting spaces optimized for use by a small number of Hosts and a large number of Guests. For more information, see What's the difference between a Virtual Meeting Room and a Virtual Auditorium?.
- Virtual Receptions: these provide a way for conference participants who cannot dial Virtual Meeting Room and Virtual Auditorium aliases directly, to access these services from a central point using DTMF tones.

Conference participants access these services by dialing one of the service's aliases. You can change the audio and video prompts presented to participants when they are accessing these services by applying themes.

In addition, the Pexip Infinity platform offers:

- Pexip Distributed Gateway: a service that allows endpoints within your deployment to make direct calls to other endpoints that use different protocols and media formats.
- Test Call Service: Pexip Infinity provides a test loopback service that allows users to check the quality of their video and audio (i.e. that their local camera, microphone and speakers are working properly), and verifies that they can connect to a Conferencing Node.

What's the difference between a Virtual Meeting Room and a Virtual Auditorium?

Virtual Auditoriums differ from Virtual Meeting Rooms in that:

- Administrators can configure a Virtual Auditorium so that Hosts and Guests see different layouts. In a Virtual Meeting Room, all participants see the same layout.
- In a Virtual Auditorium, Guest participants cannot see other Guests - they will only see the Host(s). In a Virtual Meeting Room all participants can see both Hosts and Guests, in order of who spoke most recently.
- In a Virtual Auditorium, Host participants will see all other Hosts first in the video thumbnails (in order of which Host spoke most recently), followed by Guests (again, in order of who spoke most recently). In a Virtual Meeting Room, all participants can see all other participants, in order of who spoke most recently and regardless of whether they are Hosts or Guests.
- Administrators can configure a Virtual Auditorium so that when a presentation is being shown, the person showing the presentation is fixed in the main speaker position. In a Virtual Meeting Room, this option is not available; the main speaker position will always be voice-switched, showing the current speaker.

Changing from a Virtual Meeting Room to a Virtual Auditorium and vice versa

While it is not possible to change an existing Virtual Meeting Room to a Virtual Auditorium (and vice versa), you can change the service to which an alias is routed. For example, if your sales team already have a Virtual Meeting Room with an alias **meet.sales@example.com** and you want these conferences to take advantage of the features available to a Virtual Auditorium instead, then:

1. Set up a new Virtual Auditorium (**Service Configuration > Virtual Auditoriums > Add Virtual Auditorium**) with the appropriate configuration, but do not configure any aliases.
2. **Save** the new Virtual Auditorium.
3. Go to the **Aliases** page (**Service Configuration > Aliases**) and select the alias of the existing Virtual Meeting Room.
4. From the **Service name** drop-down list, select the name of the new Virtual Auditorium.
5. Repeat for all aliases belonging to the existing Virtual Meeting Room.

All calls made to any of the aliases that previously belonged to the Virtual Meeting Room will now be routed to the Virtual Auditorium. The Virtual Meeting Room will still exist, but will not have any aliases, so no calls can be made to it.

About Virtual Meeting Rooms

A Virtual Meeting Room (VMR) is a personal virtual meeting space that is always available for use. It can host any number of people from any type of device.

Each VMR has one or more aliases associated with it. Participants access the VMR by dialing any one of its aliases - this will route them all to the same conference. For more information, see About aliases and Configuring Virtual Meeting Rooms.

Participants can access VMRs from any video endpoint, or by using one of Pexip's Infinity Connect clients — for more information see Introduction to Infinity Connect.

VMRs can be PIN-protected, and you can also assign Host and Guest privileges to conference participants. For more information, see About PINs, Hosts and Guests.

You can also place a limit on the number of participants that can access a VMR at any one time.

Administrators manage VMRs from the Pexip Infinity Administrator interface. For more information, see Configuring Virtual Meeting Rooms.

There is no limit on the number of VMRs that can be configured on your Pexip Infinity platform. VMRs do not consume resources on your deployment unless they are actually being used to host a conference. Unless you manually restrict access, the number of participants who can access a particular VMR, and the number of VMRs that can be in use at the same time, are limited only by the size of your Pexip Infinity deployment (in terms of server capacity and licenses).

VMRs can be bulk-provisioned from directory information contained in a Windows Active Directory LDAP server, or any other LDAP-accessible database.

Configuring Virtual Meeting Rooms

Virtual Meeting Rooms (VMRs) are used to hold Pexip Infinity conferences. Each VMR has one or more aliases associated with it, and participants access the conference by dialing one of these aliases.

When creating a new VMR, you must provide at least one alias that will be used to access it. You can also optionally configure PINs for the conference.

To create a new VMR, go to **Service Configuration > Virtual Meeting Rooms** and click **Add Virtual Meeting Room**.

To edit an existing VMR, or to view its details (including all the aliases associated with it), go to **Service Configuration > Virtual Meeting Rooms** and select the name of the VMR.

As an alternative to manually configuring your VMRs:

- You can import Virtual Meeting Rooms from directory information contained in a Windows Active Directory LDAP server, or any other LDAP-accessible database. For more information, see Provisioning VMRs and devices from Active Directory via LDAP.
- You can bulk import basic VMR configuration from a CSV file. See Bulk import/export of service configuration.
- You can also manage VMRs using the Pexip Infinity API. For more information, see the **Pexip Infinity Management API** documentation.

Any configuration changes made to VMRs are replicated to all Conferencing Nodes within 60 seconds and applied to any subsequent conferences in that VMR. If there are any conferences already in place that use the VMR, any attempts to join it after the configuration has been replicated may be affected by the new configuration settings. Therefore, we do not recommend changing VMR configuration while a conference is in progress as this may lead to an inconsistent user experience.

When adding or editing Virtual Meeting Rooms, the options are:

Option	Description
Name	The name used to refer to this VMR.
Description	A description of the VMR.
Creation time	This read-only field shows the date and time at which this record was first configured.

Option	Description
Host PIN	This optional field allows you to set a secure access code that must be entered by VMR participants before they can join the conference. If **Allow Guests** is set to ***Yes***, then the **Host PIN** will apply to the conference Host(s) only. For more information, see About PINs, Hosts and Guests. • PINs must use the digits **0-9** only. • PINs may optionally end with **#**. • PINs must be between 4–20 digits long, including any **#**.
Allow Guests	***Yes***: the conference will have two types of participants, Hosts and Guests. You must configure a **Host PIN** to be used by the Hosts. You can optionally configure a **Guest PIN**; if you do not configure a **Guest PIN**, Guests can join without a PIN, but the meeting will not start until the first Host has joined. ***No***: all participants will have Host privileges. Default: ***No***.
Guest PIN	This optional field allows you to set a secure access code that must be entered by VMR Guests before they can join the conference. For more information, see About PINs, Hosts and Guests. • Host PINs and Guest PINs must be different. • PINs must use the digits **0-9** only. • PINs may optionally end with **#**. • PINs must be between 4–20 digits long, including any **#**. • If the Host PIN ends in **#** and a Guest PIN is used, the Guest PIN must also end with **#**. • If **#** is not used, Host PINs and Guest PINs must have the same number of digits. • You cannot configure a Guest PIN unless you have already configured a Host PIN.
View	The maximum number of other participants that each participant will see, and the layout used to show them. For more information, see Selecting the layout seen by participants. Default: ***Large main speaker and up to 7 other participants (1+7 layout)***.
Show names of speakers	If enabled, the display name or alias of each main speaker is shown. For more information, see Showing the names of speakers. Default: ***No***.
Theme	The theme for use with this VMR. For more information, see Customizing images and voice prompts using themes. Default: ***<use Default theme>*** (the global default theme will be used).
Advanced options	
Automatically Dialed Participants	When a conference begins in this VMR, a call will be placed automatically to any participants selected here. To add an Automatically Dialed Participant that is not already on the list, select the ✚ icon to the right of the selection fields. For more information, see Automatically dialing out to a participant from a conference.
Maximum inbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being received by Pexip Infinity from each individual participant dialed in to this VMR. For more information see Restricting call bandwidth.
Maximum outbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being sent from Pexip Infinity to each individual participant dialed in to this VMR. For more information see Restricting call bandwidth.
Conference capabilities	Allows you to limit the media content of the conference. For more information, see Controlling media capability. Default: ***Main video + presentation***.

Option	Description
Participant limit	This optional field allows you to limit the number of participants allowed to join this VMR. For more information see Limiting the number of participants.
Service tag	This optional field allows you to assign a unique identifier to this service, which you can then use to track use of the service. For more information, see Tracking usage with a service tag.
VMR origin	The name of the LDAP sync template used to create this VMR; if the VMR was created by manual input, CSV import, or via the API this will be blank. This field is read-only.
Owner's email address	The email address of the owner of the VMR. VMR provisioning messages associated with this VMR will be sent to this address.
Aliases	
Alias: #1	
Alias	The alias that, when received by Pexip Infinity, will cause it to route the call to this service (Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service). The alias entered here must match exactly the alias as it is received by Pexip Infinity. Wildcards and regular expressions are not supported. In most cases, the alias received by Pexip Infinity will be the same as the alias that the conference participant dialed from their endpoint, but there are some exceptions, described in About aliases. You may also want to define multiple aliases for the same service to ensure that it can be accessed by devices and protocols that enforce specific alias formats — for more information, see Using multiple aliases to access the same service.
Description	An optional description of the alias. This is useful if you have more than one alias for a service.
Add another Alias	Select this option if you want the VMR to be accessible by more than one alias. For more information, see Using multiple aliases to access the same service.

Provisioning VMRs and devices from Active Directory via LDAP

Pexip Infinity Virtual Meeting Rooms and device aliases can be bulk-provisioned from directory information contained in a Windows Active Directory LDAP server, or any other LDAP-accessible database.

This means, for example, that you can automatically provide a personal VMR for every employee in an organization. Data such as employee names can be imported from the directory and used to generate a unique name and alias for each VMR, following a pattern such as **meet.<username>@example.com**. Other VMR settings such as PIN numbers, the layout and theme can also be assigned, depending upon your VMR template configuration.

You can also automatically provision Pexip Infinity with the device aliases that people can use to register their endpoint or Infinity Connect client to a Conferencing Node.

Each sync template can be configured to automatically synchronize against its LDAP sync source once per day. This ensures that the Pexip Infinity VMR and device configuration stays in step with all additions and removals that have occurred in the source directory. Many of the VMR / device settings can be tagged as "overridable" which means that if that setting (the VMR PIN number for example) is manually overridden after the VMR or device has been created, a subsequent resynchronization will not reset it back to another value.

If Pexip Infinity has been configured with the details of an SMTP server, you can send an email to the VMR owner whenever a synchronization creates a new VMR or modifies an existing VMR, or to inform a user of the device alias and credentials that they need to use to register their device.

Configuration summary

To provision your Virtual Meeting Rooms or device aliases from an LDAP database you must:

1. Configure the connection details of an LDAP data source. You must also ensure that Pexip Infinity has trusted CA certificates for the authority that signed the LDAP server's certificate (if a TLS connection is required).

2. Configure an LDAP synchronization template that controls what to synchronize — VMRs, device aliases or both — and how individual settings such as the VMR name, dialable alias and PIN numbers are generated, which of those values can be manually overridden, and whether to synchronize automatically against the LDAP data source.

   If required, you can configure multiple synchronization templates that use the same LDAP data source (or different templates that use different LDAP sources). For example, you may want to use a different template for VMR syncing than the template used for device alias syncing if you need to use different LDAP filter criteria for those two user bases.
3. Configure an SMTP server and construct an email template, if you want to send a provisioning notification email to the VMR owner or intended user of a device alias.
4. Generate the VMRs and device aliases using your template and LDAP data source. Set the template to sync automatically when you are happy that the template is configured correctly.

These configuration steps are described in more detail in the following sections.

In addition, you can also:

- View the status of ongoing or completed LDAP template synchronization processes via **Status > LDAP Sync Status**.
- Delete all of the VMRs and devices that have been created from a specific template.

Configuring an LDAP data source

To configure Pexip Infinity with the connection details of an LDAP data source, such as a Windows Active Directory LDAP server:

1. Go to **Utilities > LDAP Sync Sources**.
2. Select **Add LDAP sync source**, and then complete the following fields:

Option	Description
Name	The name used to identify this LDAP sync source.
Description	An optional description of the sync source.
LDAP server address	The domain name (for DNS SRV lookup), FQDN (for DNS A/AAAA lookup) or IP address of the LDAP server. If using a domain or an FQDN, ensure that it is resolvable over DNS. You must also ensure that Pexip Infinity has trusted CA certificates for the authority that signed the LDAP server's certificate (if a TLS connection is required). We strongly recommend that you do not use an IP address. If an IP address is used, and a TLS connection is required, this will only work if the IP address is specified as the common name in the LDAP server's certificate. See Troubleshooting LDAP server connections for more information about how the system establishes a connection to the LDAP server and how to troubleshoot connection issues.
Search global catalog	Select this option to expand the scope of the search to the entire Active Directory Global Catalog instead of traditional LDAP.
Allow insecure transport	By default the system will attempt to establish a secure TLS connection with the LDAP server. Select this option if you want to allow the system to fall back to a TCP connection (using SASL DIGEST-MD5). You cannot specify the LDAP server by IP address if this option is selected.
LDAP bind username and password	The username and password of the bind account on the LDAP server. This should be a domain user service account, not the Administrator account.
LDAP base DN	The base DN (distinguished name) of the LDAP forest to query (e.g. **dc=example,dc=com**).

3. Select **Save**.

   You can save the sync source details only if Pexip Infinity can successfully contact the specified LDAP server.

Note that all LDAP distinguished names must be entered as per the LDAP standard (**RFC 4514**). LDAP configuration is case insensitive.

Configuring an LDAP synchronization template

LDAP synchronization templates control what to synchronize — VMRs, device aliases or both — and how individual settings such as the VMR name, dialable alias and PIN numbers are generated, and what to include in provisioning notification emails sent to the VMR or device owner. You must also tell the template which LDAP data source to use.

To configure Pexip Infinity with an LDAP synchronization template:

1. Go to **Utilities > LDAP Sync Templates**.
2. Select **Add LDAP sync template**, and then complete the following fields:

Option	Description
Name	The name used to identify this synchronization template.
Description	An optional description of the synchronization template.
LDAP user search DN	The DN relative to the **LDAP base DN** of the sync source to query for user records (e.g. **ou=people**). If blank, the **LDAP base DN** is used. In deployments with large user bases, you may want to configure this to optimize the LDAP user queries.
LDAP user filter	The LDAP filter used to match user records in the directory. Default: (&(objectclass=person)(!(objectclass=computer))) For more information, see LDAP search and filter examples.
LDAP sync source	Select the LDAP data source to use when syncing records. Note that you can select ✚ which will open a new window from where you can configure a new sync source.
Sync VMRs	Enables VMR synchronization for this template. Default: enabled.
Sync devices	Enables device alias synchronization for this template. Default: disabled.
Enable automatic daily sync	Select this option to instruct Pexip Infinity to automatically synchronize this template against its LDAP sync source once per day. This ensures that VMRs are regularly updated, deleted or created as appropriate based on the latest data in the sync source. Therefore, for example, if a user is removed from Active Directory or their account is disabled, their corresponding VMR will be deleted via the next daily sync. All automatic synchronizations are initiated at 01:00 UTC (this start time cannot be configured). As template synchronization can result in the automatic creation, modification or deletion of large numbers of VMRs and devices, we recommend that you only enable automatic syncing after you have manually synced at least once and have verified that you are satisfied with your sync template configuration.
Device settings (shown when Sync devices is selected)	
Device alias pattern	The pattern for the alias that will be registered by the device and be used by people trying to call the device. For more information, see Using patterns to format the VMR and device properties (name, alias etc). Default: {{mail}} If a device with the generated alias already exists, that existing device configuration will be overwritten with the data generated from this template.
Device tag pattern	The pattern for the unique identifier used to track usage of the device.
Allow device tag to be manually overridden	Allows the auto-generated device tag to be manually overridden for each device alias.
Device description pattern	The pattern to use when generating the optional description of this device alias.

Option	Description
Allow device description to be manually overridden	Allows the auto-generated device description to be manually overridden for each device alias.
Device username pattern	The pattern for the device username. Note that you can use the same username for different device aliases (although we recommend you only do this for multiple devices used by the same person). Default: {{mail}}
Allow username to be manually overridden	Allows the username to be manually overridden for each device alias.
Device password pattern	The pattern to use when generating the password for this device. A password pattern must be specified if a username pattern is configured. Example: **{{ ("some random string"+(mail|pex_reverse))|pex_hash|pex_tail(16) }}**
Allow password to be manually overridden	Allows the password to be manually overridden (by the administrator) for each device alias.
VMR settings (shown when Sync VMRs is selected)	
VMR name pattern	The pattern to use to generate the name of the VMR. You should structure this pattern to generate a unique VMR name. For more information, see Using patterns to format the VMR and device properties (name, alias etc). Example: **{{givenName}} {{sn}} VMR** If a VMR with the generated name already exists, that existing VMR configuration will be overwritten with the data generated from this template.
VMR description pattern	The pattern to use to generate the VMR description. This field is optional. Examples: **{{givenName}} {{sn}}'s meeting room** You could use a more advanced pattern such as: **{{givenName}} {{sn}}'s {%if department %} ({{department}}){% endif %} meeting room** which will insert "(<department name>)" before "meeting room", but only if a value exists in the LDAP **department** field.
Allow VMR description to be manually overridden	Allows the auto-generated VMR description to be manually overridden for each VMR.
Host PIN pattern	The pattern to use to generate the Host PIN (if required). Example: **{{ [9,pex_random_pin(5)]|join }}** (sets the PIN to a random 6 digit number starting with a 9) In this example you would probably also select **Allow PIN settings to be manually overridden** to stop the PIN being changed to another random value after every template resync.
Allow Guests	***Yes***: the conference will have two types of participants, Hosts and Guests. You must configure a **Host PIN** to be used by the Hosts. You can optionally configure a **Guest PIN**; if you do not configure a **Guest PIN**, Guests can join without a PIN, but the meeting will not start until the first Host has joined. ***No***: all participants will have Host privileges. Default: ***No***.
Guest PIN pattern	The pattern to use to generate the Guest PIN (if required). Example: **{{ [2,pex_random_pin(5)]|join }}** (sets the PIN to a random 6 digit number starting with a 2) Note that if you set a Host PIN and a Guest PIN for a VMR, those two PINs must be different. Therefore we recommend that you generate Host and Guest PINs of different lengths, or ensure that some aspect of each PIN will be different (as shown in the examples).

Option	Description
Allow PIN settings to be manually overridden	Allows the auto-generated Host and Guest PIN settings to be manually overridden for each VMR. Note that this would also preserve the generated PIN value, even if it does not actually get manually overridden.
Layout	The maximum number of other participants that each participant will see, and the layout used to show them. For more information, see Selecting the layout seen by participants. Default: ***Large main speaker and up to 7 other participants (1+7 layout)***.
Show names of speakers	If enabled, the display name or alias of each main speaker is shown. For more information, see Showing the names of speakers.
Allow layout settings to be manually overridden	Allows the type of layout and whether to show names of speakers to be manually overridden for each VMR.
Theme	The theme for use with this VMR. For more information, see Customizing images and voice prompts using themes. Default: ***<use Default theme>*** (the global default theme will be used).
Allow theme to be manually overridden	Allows the auto-generated theme to be manually overridden for each VMR.
VMR alias 1 pattern	The pattern to use to generate the alias (string) that participants can dial to join this VMR. You should structure this pattern to generate a unique alias for the VMR. Example: **meet.{{givenName|lower}}.{{sn|lower}}@example.com** or **meet.{{mail}}** If the generated alias is already assigned to another VMR, that alias will be reassigned to the VMR currently being created/updated from this template.
Alias 1 description pattern	The pattern to use to generate a description of the alias. This field is optional. Example: **{{givenName|lower}}'s full URI**
VMR alias 2...4 pattern and descriptions	You can enter optionally enter patterns for a further 3 alternative aliases, such as an E.164 alias, to associate with this VMR. Note: do not use the pex_random_pin() filter to generate numeric aliases as this is likely to generate duplicate aliases. Where possible, it is best to use a phone number, employee ID or other typically unique value as the basis of a numeric alias. If a unique value does not exist, a good fallback is to use a pex_hash() expression such as the following to generate a consistent 6-digit random E164-type alias from a hash of each user's email address: **{{ ("my alias passphrase"+(mail|pex_reverse))|pex_hash|pex_tail(6) }}**
Email options	
Send emails VMR / device owner's email address Address override SMTP server	These email options allow you to send an email to the VMR owner whenever a synchronization creates a new VMR or modifies an existing VMR, or when a synchronization creates a new device alias or modifies an existing device alias. See Sending provisioning emails to VMR and device owners for information about how to configure these options.
Advanced options (shown when Sync VMRs is selected)	
Maximum inbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being received by Pexip Infinity from each individual participant dialed in to this VMR. For more information see Restricting call bandwidth.
Maximum outbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being sent from Pexip Infinity to each individual participant dialed in to this VMR. For more information see Restricting call bandwidth.
Allow maximum bandwidth settings to be manually overridden	Allows the auto-generated maximum inbound call bandwidth and outbound call bandwidth to be manually overridden for each VMR.

Option	Description
Participant limit	This optional field allows you to limit the number of participants allowed to join this VMR. For more information see Limiting the number of participants.
Allow participant limit to be manually overridden	Allows the auto-generated participant limit to be manually overridden for each VMR.
Service tag pattern	This optional field allows you to assign a unique identifier to this service, which you can then use to track use of the service. For more information, see Tracking usage with a service tag.
Allow service tag to be manually overridden	Allows the auto-generated service tag to be manually overridden for each VMR.
Conference capabilities	Allows you to limit the media content of the conference. For more information, see Controlling media capability. Default: ***Main video + presentation***.

3. Select **Save**.
   You must save the template before you can use it to generate VMRs.

Allowing generated fields to be manually overridden

Many of the VMR settings can be tagged as "overridable" which means that if that setting is manually overridden after the VMR has been created, a subsequent resynchronization of that template will not reset it back to another value.

For example, you could configure the template with **Allow PIN settings to be manually overridden** selected. Then, after generating a VMR with a random Host PIN, the administrator could manually change the PIN to another specific value. This "overridable" setting ensures that the new value is protected and is not reset to another random value when that template is next used to synchronize the VMRs.

To make a field modifiable via the template again, you must clear the relevant **Allow <field> to be manually overridden** template setting and then re-sync with that template. Note that a VMR setting tagged as overridable via one template could still be modified by a different template.

LDAP search and filter examples

You can configure multiple synchronization templates that all use the same LDAP data source (or different LDAP sources, if required), but that apply different filters to that LDAP source. This means that you could apply different VMR configuration patterns based on different LDAP organizational groups. For example, you could configure the VMRs for all members of the European sales team to have a Guest PIN and a different theme to those VMRs generated for the American accounting department.

If you want to apply different VMR patterns to different types of users, you can use the **LDAP user search DN** and **LDAP user filter** template fields to filter the records from the LDAP sync source that are used to generate VMRs via that template.

This section contains some examples that you could use in a Windows Active Directory environment. Note that all LDAP user search and user filter contents are not case sensitive.

Filtering based on group membership

To import users that belong to a specific group, you can filter on the **memberOf** AD attribute. For example, to only import users who are members of the cn=sales,ou=groups,dc=example,dc=com group, you would set the **LDAP user filter** on your template to:
(&(objectCategory=person)(objectClass=user) (memberOf=cn=sales,ou=groups,dc=example,dc=com))

You could then configure a second template for the accounting department that uses the same **LDAP sync source** but with the **LDAP user filter** set to:
(&(objectCategory=person)(objectClass=user) (memberOf=cn=accounting,ou=groups,dc=example,dc=com))

To only import users who are **not** a member of the sales group, you would set the **LDAP user filter** to:
(&(objectCategory=person)(objectClass=user) (!(memberOf=cn=sales,ou=groups,dc=example,dc=com)))

Filtering by LDAP field names

To import only those users who have an email address, you can set the **LDAP user filter** to:
mail=*

To import all users **except** for some named individuals, you could use the following **LDAP user filter** model:
(&(objectCategory=person)(objectClass=user) (!(cn=Alice Parkes))(!(cn=Bob Lee))(!(cn=Carol Jones)))

Filtering by organizational unit (e.g. region/country)

To import users that are located in a specific organizational unit such as a region or country, you can restrict the user search to the appropriate 'ou' objects. For example, to only import users who are based in 'Europe', you would set the **LDAP user search DN** to:
ou=europe,ou=people
(this is the DN relative to the base DN defined in the LDAP sync source)

More information on Active Directory LDAP filtering can be found at http://social.technet.microsoft.com/wiki/contents/articles/5392.active-directory-ldap-syntax-filters.aspx.

Using patterns to format the VMR and device properties (name, alias etc)

When you configure an LDAP synchronization template, you define patterns for how some of the VMR or device properties — such as its name and the aliases — are generated from the data in the LDAP sync source.

Pexip Infinity's pattern formats are based on a subset of the jinja2 templating language (http://jinja.pocoo.org/docs/dev/).

These patterns are made up of the following elements:

- **literal text**, such as prefixing every name with **meet.**
- **variables** that are substituted with values from the LDAP sync source, such as **givenName**
- **filters** that can manipulate text or modify the content of variables or text strings, such as **join** or **lower**
- **delimiters** such as **{{...}}** and pipes **|** which are used to enclose variables and define filter expressions
- **jinja statements and control structures** (see http://jinja.pocoo.org/docs/dev/templates/#list-of-control-structures)

An example pattern would be **meet.{{givenName|lower}}.{{sn|lower}}**. This concatenates the literal text **meet.** with the content of the **givenName** variable that has been converted to lower case by the **lower** filter. This is then concatenated with a period **.** and then the **sn** (surname) variable, also piped through the **lower** filter.

Therefore if Alice Parkes had an entry in the LDAP directory with **givenName=Alice** and **sn=Parkes**, the example pattern above would produce **meet.alice.parkes**.

The following sections provide more information about the supported variables and filters, and how to format expressions.

Supported variables

When syncing each VMR, Pexip Infinity extracts LDAP fields from each user entry in the LDAP data source and makes them available as variables (with the same name) for use in the template. Pexip Infinity offers several standard fields, and administrators can also add their own custom set of fields.

When using variables:

- You must enclose the variable name within **{{...}}** delimiters, for example **{{**givenName**}}**.
- Any combination of the variables can be used in any VMR synchronization template pattern field.
- Fields that are not present in the LDAP source will appear as "None".
- Capitalization of the variable name is important. The variable **must** be spelled exactly as shown below.

Standard LDAP fields

Pexip Infinity automatically extracts and makes available the following standard LDAP fields:

Variable name	Description
company	Company/organization name
department	Department name
displayName	User's preferred display name
employeeID *	Employee reference number
givenName	First name
mail	Email address
mailNickname *	Typically used to store an alternative email address (e.g. based on a maiden name); see Change of name below
mobile	Mobile phone number
objectGUID	Globally unique identifier for the directory object (user entry)
sAMAccountName	Logon name (domainname\username format)
sn	Last name or surname
telephoneNumber	Telephone number
title	Job title
userPrincipalName	Logon name (username@domainname format)

* not present in all AD schemas

Custom LDAP fields

In addition to the standard set of fields made available automatically by Pexip Infinity, administrators can configure their own custom set of fields to support additional attributes in their LDAP/AD schemas.

To add a custom LDAP field:

1. Go to **Utilities > LDAP Sync Fields**.
2. Select **Add LDAP sync field**, and then complete the following fields:

LDAP field name	The name of LDAP field to be read from the LDAP server. The name is case sensitive. If the name does not match exactly against a field in the LDAP data source its value will appear (when used in a template) as "None".
Template variable name	The name of the variable to use in VMR sync templates that will contain the value of the LDAP field. We recommend that you set this name to something similar to the LDAP field name, but note that this name cannot contain hyphens.
Description	An optional description of the LDAP attribute.
Is binary (advanced options)	In advanced scenarios, some binary LDAP fields such as GUIDs require special encoding. In such cases, expand the advanced options and select **Is binary**. Do not select this option for ordinary textual or numeric LDAP fields.

3. Select **Save**.
   The **Template variable name** that represents the custom LDAP field can now be used in a template along with the standard LDAP fields.

Change of name

A typical scenario encountered by IT administrators is when someone changes their name (e.g. after getting married) and wants to preserve their original contact address alongside their new contact details.

One way in which you could support this in Pexip Infinity is by defining multiple VMR alias patterns in your template. This example approach assumes that the LDAP **mail** field holds the user's contact address, and a **mailNickname** field is used to hold an alternative address. Thus, you could define the following patterns:

VMR alias 1 pattern:	meet.{{mail}}
VMR alias 2 pattern:	{%if mailNickname %}{#Add alias based on user's maiden name mailNickname#}meet.{{mailNickname}}{% else %}{#intentionally leave the alias blank#}{% endif %}

For most users, this will generate a single VMR alias in the format **meet.<email address>**, and the VMR's second alias will remain blank as the LDAP **mailNickname** field for those users will be empty.

For example, user Ann Jones has her LDAP **mail** field set to ann.jones@example.com and her LDAP **mailNickname** field is blank. Her VMR will have a single alias of **meet.ann.jones@example.com**.

When a user changes their name, the administrator would update the user's LDAP **mail** field to match their new name and populate the **mailNickname** field with the previous value of the **mail** field.

Now, the next time the administrator performs a template synchronization, the user's VMR aliases will be updated. The first alias will be changed to a new string based on the user's new email address, and a second alias will also be generated based upon the **mailNickname** field. This means that the user can be contacted via either their previous VMR alias or the new alias.

In our example, let's assume that Ann Jones changes her name to Ann Smith. The administrator changes her LDAP **mail** field to ann.smith@example.com and sets her LDAP **mailNickname** field to ann.jones@example.com. Her VMR will now have two aliases: **meet.ann.smith@example.com** and **meet.ann.jones@example.com**.

Supported filters

Pexip Infinity supports a subset of filters from the jinja2 templating language. Any jinja filters that are not listed below have been disabled in Pexip Infinity. See http://jinja.pocoo.org/docs/dev/templates/#filters for more information about these filters.

abs	float	last	striptags
capitalize	format	lower	trim
default	int	replace	truncate
first	join	round	upper

To use a filter you would typically follow the syntax **{{<source_value>|<filter_name>}}**.

In most cases the <source_value> is likely to be a variable, for example **{{givenName|upper}}**, although it could be one or more literal values, for example **{{ [1, 2, 3, 4]|join }}**.

Some filters take parameters, for example **{{sn|truncate(5)}}**. You can also use multiple filters in the same expression, for example **{{sn|truncate(5)|upper}}**.

The **replace** filter is often used. This replaces one string with another string. The first argument of the filter is the substring that should be replaced, the second is the replacement string. If the optional third argument count is given, only the first count occurrences are replaced.

Example usage: **{{ department|replace("Personnel", "HR") }}**

If the department field contained "Personnel Department Personnel", this would be converted to "HR Department HR".

Example usage: **{{ department|replace("Personnel", "HR", 1) }}**

In this case a count of 1 is specified, thus if the department field contained "Personnel Department Personnel", it would be converted to "HR Department Personnel".

For more complicated search and replace patterns, use the custom Pexip **pex_regex_replace** filter described below.

Custom Pexip filters

In addition to the jinja filters, Pexip also provides the following custom filters:

pex_regex_replace('find_regex', 'replace_string')

This performs a regex find and replace.

Example usage: **{{ givenName|pex_regex_replace('(e|o)', 'i') }}**

This example replaces occurrences of 'e' or 'o' with 'i' in the givenName variable.

See Regular expression (regex) reference for information about writing regular expressions.

pex_reverse

This reverses the characters in the input field.

Example usage: **{{ givenName|pex_reverse|lower }}**

In this example, if givenName is 'Alice', this expression would return 'ecila'.

pex_clean_phone_number

This extracts only +0123456789 characters (and removes ()&%#@|"':;, A-Z,a-z etc).

Example usage: **{{ telephoneNumber|pex_clean_phone_number }}**

In this example, if telephoneNumber is '+44 (20) 12345678', this expression would return '+442012345678'.

pex_strlen

This returns the length of string.

Example usage: **{%set a = telephoneNumber|pex_clean_phone_number|pex_head(8)%}{%set b = mobile|pex_clean_phone_number|pex_head(8)%}{%if a|pex_strlen== 8%}{{a}}{%else%}{{b}}{%endif%}**

In this example, the expression returns the value of the telephoneNumber field if it is present and at least 8 characters long after cleaning and truncation, otherwise it returns the value of the mobile field that has been cleaned and truncated to at most 8 characters.

Example usage: **{%if telephoneNumber|pex_strlen == 5 %}{#return user's phone number#}{{telephoneNumber|pex_clean_phone_number}}{% else %}{#intentionally leave blank#}{% endif %}**

In this example, the expression returns a cleaned value of the telephoneNumber field if it is 5 characters long, otherwise it returns an empty string.

pex_require_min_length(length)

You can use this filter to control if VMRs are created or not, based on the length of a string or variable.

For example, you could set the **VMR description pattern** as **{{ mail|pex_require_min_length(1) }}**. This means that the VMR will not be created if the mail variable is empty. (When subsequently syncing, the VMR would be deleted if the condition is not met.)

Example usage: **{{ telephoneNumber|pex_require_min_length(4) }}**

This example ensures that a VMR is created only if telephoneNumber is at least 4 characters long.

pex_random_pin(length)

Generates a random PIN of the given length. Note that this filter does not take any input.

Example usage: **{{ [9,pex_random_pin(5)]|join }}** for the host PIN, and **{{ [2,pex_random_pin(5)]|join }}** for the guest PIN.

A usage pattern such as this (where the host PIN starts with a 9, and the guest PIN starts with a 2) ensures that a VMR can never be configured with a host PIN that is the same as the guest PIN.

You would typically use this filter in conjunction with the **Allow PIN settings to be manually overridden** option to ensure that the PIN is not reset every time a template resync is performed.

Do not use pex_random_pin() to generate aliases. This filter generates a truly random number, with each number generated independently of any previous output. Therefore, with many thousands of users, a 5 digit numeric alias (e.g. pex_random_pin(5)) is statistically quite likely to clash with a random alias generated using pex_random_pin(5) for another user. With 10,001 or more users and a 4 digit random alias, a clash is guaranteed (with multiple clashes being likely).

pex_uuid4()

This generates a uuid (universally unique identifier). Note that this filter does not take any input.

Example usage: **{{ pex_uuid4() }}**

In a similar manner to the **pex_random_pin** filter, you would typically use this in conjunction with the appropriate **Allow <field> to be manually overridden** template setting, to ensure that after a uuid has been assigned to a field, another different uuid is not assigned every time a template resync is performed.

pex_hash

Performs a hash of a field. You could use this, for example, to generate random-looking PINs based on a hash of one or more fields from the directory (you would also need to apply another filter such as pex_tail to set the PIN to an appropriate valid length).

Example usage: **{{ sAMAccountName|pex_hash|pex_tail(6) }}**

pex_head(maxlength)

Returns, at most, the first maxlength characters from the input field.

Example usage: **{{ givenName|pex_head(4) }}**

In this example, for a givenName of 'Alice' this expression would return 'Alic', and a givenName of 'Bob' would return 'Bob'.

pex_tail(maxlength)

Returns, at most, the last maxlength characters from the input field.

Example usage: **{{ telephoneNumber|pex_tail(4) }}**

In this example, if telephoneNumber is '+44 (20) 12345678', this expression would return '5678'.

pex_base64

Performs Base64 encoding on the input field.

Example usage: **{{provisiondata|pex_base64}}**

In this example, a local variable named **provisiondata** is encoded as Base64.

Generating / syncing VMRs and devices with LDAP

After you have configured a sync source and a template you can generate your VMRs and devices for the first time, or you can synchronize them to any changes that have occurred in the LDAP database since the last time the VMRs and devices were updated.

When synchronizing a template:

- Pexip Infinity checks all of the VMRs and devices previously created by that template and it updates, deletes or creates new VMRs / devices as appropriate based on the current data in the sync source. Therefore, for example, if a user is removed from Active Directory or their account is disabled, then their corresponding VMR and device alias will be deleted.
- If a generated VMR name or device alias already exists (either created manually or via a different template), that existing VMR or device alias will be updated with the new settings generated from the template.
- If a generated alias for a VMR is already assigned to another VMR, that alias will be reassigned to the VMR currently being created/updated.

Any configuration changes made to Virtual Meeting Rooms are replicated to all Conferencing Nodes within 60 seconds and applied to any subsequent conferences in that Virtual Meeting Room. If there are any conferences already in place that use the Virtual Meeting Room, any attempts to join it after the configuration has been replicated may be affected by the new configuration settings.

Changes to device aliases are also replicated to all Conferencing Nodes within 60 seconds and applied to any subsequent registration requests.

The template synchronization process can take several minutes if you have a large number of VMRs or devices. Therefore you must wait for 60 seconds after the entire synchronization process has completed to ensure that all synced data has been replicated to all Conferencing Nodes.

For these reasons, we do not recommend changing Virtual Meeting Room configuration while a conference is in progress as this may lead to an inconsistent user experience.

Automatic synchronization

To automatically synchronize a template on a daily basis:

1. Go to **Utilities > LDAP Sync Templates**.
2. Select the template you want to use (to open the **Change LDAP Sync Template** page).
3. Select **Enable automatic daily sync**.
4. Select **Save**.

All automatic synchronizations are initiated at 01:00 UTC (this start time cannot be configured).

As template synchronization can result in the automatic creation, modification or deletion of large numbers of VMRs and devices, we recommend that you only enable automatic syncing after you have manually synced at least once and have verified that you are satisfied with your sync template configuration.

Manual synchronization

To manually generate (for the first time) or synchronize the VMRs and devices in Pexip Infinity with the LDAP data source:

1. Go to **Utilities > LDAP Sync Templates**.
2. Select the template you want to use (to open the **Change LDAP Sync Template** page).
3. Scroll down to the bottom of the page and select **Sync now**.
4. A status page is displayed which shows the progress of the synchronization.
   You can safely navigate away from this page without affecting the synchronization.
5. For large imports that take a long time to complete, you can monitor the ongoing status of the import at any time via **Status > LDAP Sync Status**.

Alternatively, you can sync one or more templates by going to **Utilities > LDAP Sync Templates**, selecting the check box next to the templates that you want to use, and then from the **Action** drop-down menu, select ***Sync selected LDAP sync templates*** and then select **Go**.

Deleting all VMRs and devices associated with a template

You can easily delete all of the VMRs and devices that have been created from a specific template, for example if you have incorrectly specified the template's **LDAP user filter**.

To delete all VMRS and devices associated with a synchronization template:

1. Go to **Utilities > LDAP Sync Templates**.
2. Select the template whose associated VMRs and devices you want to delete (to open the **Change LDAP Sync Template** page).
3. Scroll down to the bottom of the page and select **Delete all VMRs and devices created from this template**.
4. Confirm that you want to proceed.
5. The VMRs and devices will be deleted and a status message is displayed confirming the number of VMRs and devices that were deleted.

Deleting a template

If you delete a synchronization template, all of the VMRs and devices associated with that template will also be deleted.

Sending provisioning emails to VMR and device owners

When bulk-provisioning VMRs and device aliases from Active Directory via LDAP, an email can be sent out to the VMR owner or device owner telling them the:

- VMR properties, such as its aliases and security PINs
- device alias properties, such as its associated username and password for registration purposes.

Email generation is specified on a per sync template basis. The content of the email is free-form and can be customized for each generated VMR or device alias by using variables — such as {{pin}} to include the Host PIN in VMR-related emails — that are substituted with the relevant value when each individual email is generated.

You can also send reminder emails to specific VMR or device owners, or to all VMRs and devices associated with a sync template.

Guidelines and limitations

When constructing your email template:

- Ensure that the subject line of the generated email is a single line.
- You can use HTML markup (UTF-8 characters only).

Configuring an LDAP sync template to generate emails

When configuring an LDAP sync template (**Utilities > LDAP Sync Templates**) you can specify the following email-related options:

Email options	
Send emails	When selected, the system generates and sends an email to the: • VMR / device owner whenever a synchronization creates a new VMR or modifies an existing VMR (when **Sync VMRs** is enabled for the template) • device owner when a synchronization creates a new device alias or modifies an existing device alias (when **Sync devices** is enabled for the template). Separate emails are sent for VMR provisioning and for device provisioning when both **Sync VMRs** and **Sync devices** are enabled. Note that you can also manually send reminder emails.
VMR / device owner's email address	The email address of the owner of this VMR or device alias. The generated email(s) will be sent to this address. Example: **{{mail}}**
Allow email address to be manually overridden	Allows the auto-generated email address to be manually overridden for each VMR or device alias.
SMTP server	Select the SMTP server to use for sending provisioning emails (see Configuring SMTP servers).
VMR email subject template	A template for the subject line of the email to be sent when a VMR is created or updated.
VMR email body template	A template for the body of the email to be sent when a VMR is created or updated. See Constructing the VMR provisioning email below for more information on how complete this field.
Device email subject template	A template for the subject line of the email to be sent when a device alias is created or updated.
Device email body template	A template for the body of the email to be sent when a device alias is created or updated. See Constructing the device alias provisioning email below for more information on how complete this field.

Constructing the VMR provisioning email

The **VMR email subject template** and **VMR email body template** fields contains the pattern for the email to be sent to the VMR owner when that VMR is first created (provisioned) or is updated.

The templates will typically contain a mixture of literal text and variables, and it can also contain control structures that allow you to vary the content of the email based upon certain conditions.

Supported variables

The **VMR email subject template** and **VMR email body template** fields support a different set of variables from the other synchronization template pattern fields (because it is based on the generated VMR properties, rather than the source data used to build those properties).

The following variables are available:

Variable name	Description
primary_owner_email_address	The email address of the owner of the VMR.
description	The description of the VMR.
aliases	A comma-separated list of all of the VMR's aliases, e.g. **"alice", "alice@example.com", "123456"** Note that you may also use the system-generated **uri_alias** and **numeric_alias** variables as described below.
allow_guests	Whether to distinguish between Host and Guest participants: true — the conference has two types of participants: Hosts and Guests. false — all participants have Host privileges
pin	The Host PIN.
guest_pin	The Guest PIN.
max_callrate_in	The maximum inbound call bandwidth (kbps).
max_callrate_out	The maximum outbound call bandwidth (kbps).
participant_limit	The maximum number of participants allowed to join this VMR.
tag	The VMR's service tag.
Additional generated variables	
action	Possible values are: • "created" — when the VMR has just been created • "updated" — when the existing VMR has been updated • "reminder" — when sending reminder emails.
uri_alias	Set to one of the VMR's URI-style aliases i.e. an alias in the format alias@domain. It is blank if there is no URI-style alias.
numeric_alias	Set to one of the VMR's all-numeric aliases i.e. an alias consisting entirely of digits 0-9. It is blank if there is no numeric alias.

For more information about how participants join conferences, see Creating preconfigured links to conferences via Infinity Connect.

Example VMR email templates

Here are some example email body templates that you can use as the basis for your own emails.

Basic VMR email template

This is a basic email template that can be used to show joining instructions for VMRs that do not have a PIN.

It makes use of the `{{numeric_alias}}` and `{{uri_alias}}` variables which, when the email is generated, will be substituted with the appropriate alias values for that specific VMR.

You can remove the instructions for any of the joining methods that are not appropriate for your deployment, such as the instructions for joining via the Infinity Connect client or via telephone. If you use this example, remember to replace `rp.example.com` with the address of your reverse proxy or Conferencing Node, and to use your own direct dial telephone number instead of `+1 555 0123`.

```
Here are the details of your Pexip Virtual Meeting Room (VMR). You can copy and paste
these instructions when inviting people to join your meetings.

From a web browser, go to: https://rp.example.com/webapp/#/?conference={{numeric_alias}}
then enter your Name and select "Connect"

From a video conferencing endpoint dial {{uri_alias}}

To join from an Infinity Connect client, click pexip:{{uri_alias}} (you can download it
from https://www.pexip.com/download-infinity-connect or from the app store for your
iOS/Android device)

To join from a telephone, dial +1 555 0123 then, when prompted, enter the conference
number {{numeric_alias}} followed by "#"

Instructions for using your VMR can be found at https://docs.pexip.com/admin/connect_
quick.htm
```

Basic VMR email template with Host PIN and no-PIN variations

This is an enhancement to the basic email template that uses an if... else... control structure to generate two versions of content based upon whether the VMR has a Host PIN or not.

The content following `{% if pin %}` is used if the VMR has a Host PIN, otherwise (there is no Host PIN) the content between `{% else %} {# if pin #}` and `{% endif %} {# if pin #}` is used instead. Note that the version of the instructions for PIN-protected VMRs makes use of the `{{pin}}` variable.

```
Here are the details of your Pexip Virtual Meeting Room (VMR). You can copy and paste
the "How to join" section when inviting people to join your meetings.

{% if pin %}

Your VMR is PIN-protected. How to join:

From a web browser, go to: https://rp.example.com/webapp/#/?conference={{numeric_
alias}}&pin={{pin}} then enter your Name and select "Connect"

From a video conferencing endpoint dial {{uri_alias}}: then enter the PIN: {{pin}}

To join from an Infinity Connect client, click pexip:{{uri_alias}}?pin={{pin}} (you can
download it from https://www.pexip.com/download-infinity-connect or from the app store
for your iOS/Android device)

To join from a telephone, dial +1 555 0123 then, when prompted, enter the conference
number {{numeric_alias}} followed by "#". When prompted, enter the PIN: {{pin}}

{% else %} {# if pin #}

Your VMR is not protected by a PIN. This means that anyone can use the room at any time.
How to join:

From a web browser, go to: https://rp.example.com/webapp/#/?conference={{numeric_alias}}
then enter your Name and select "Connect"

From a video conferencing endpoint dial {{uri_alias}}

To join from an Infinity Connect client, click pexip:{{uri_alias}} (you can download it
from https://www.pexip.com/download-infinity-connect or from the app store for your
iOS/Android device)

To join from a telephone, dial +1 555 0123 then, when prompted, enter the conference
number {{numeric_alias}} followed by "#"

{% endif %} {# if pin #}
```

```
Instructions for using your VMR can be found at https://docs.pexip.com/admin/connect_
quick.htm
```

Email VMR template with nested conditional text and variables for repeated text snippets

This template adds to the previous example by using nested `if... else...` control structures in the email to only send specific parts of the email based on certain conditions, such as whether the VMR has been created or updated, or whether it contains Host and Guest PINs.

It also demonstrates how you can define variables to contain items of text that you want to use many times within the email and then shows how to use those variables within the template.

```
{%set pstn_access_number = "+1 555 0123" %}

{%set rp_address = "rp.example.com" %}

{%set app_download_location = "(you can download it from https://www.pexip.com/download-
infinity-connect or from the app store for your iOS/Android device)" %}

{%if action=="created" %}Welcome - your new Pexip Virtual Meeting Room (VMR) has been
created.{% else %}Service update: Your Pexip Virtual Meeting Room (VMR) has been updated
with new settings. {%endif%}

Here are the details of your VMR. You can copy and paste the "How to join" section when
inviting people to join your meetings.

{%if allow_guests %}

How to join if you are a host:

From a web browser, go to: https://{{rp_address}}/webapp/#/?conference={{numeric_
alias}}&pin={{pin}} then enter your Name and select "Connect"

From a video conferencing endpoint dial {{uri_alias}}: then enter the PIN: {{pin}}

To join from an Infinity Connect client, click pexip:{{uri_alias}}?pin={{pin}}

To join from a telephone, dial {{pstn_access_number}} then, when prompted, enter the
conference number {{numeric_alias}} followed by "#". When asked enter the host PIN:
{{pin}}

How to join if you are a guest:

{%if guest_pin %}

From a web browser, go to: https://{{rp_address}}/webapp/#/?conference={{numeric_
alias}}&pin={{guest_pin}} then enter your Name and select "Connect"

From a video conferencing endpoint dial {{uri_alias}}: then enter the guest PIN:
{{guest_pin}}

To join from an Infinity Connect client, click pexip:{{uri_alias}}?pin={{guest_pin}}
{{app_download_location}}

To join from a telephone, dial {{pstn_access_number}} then, when prompted, enter the
conference number {{numeric_alias}} followed by "#". When prompted, enter the guest PIN:
{{guest_pin}}

{% else %} {# if guest_pin #}

From a web browser, go to: https://{{rp_address}}/webapp/#/?conference={{numeric_alias}}
then enter your Name and select "Connect"

From a video conferencing endpoint dial {{uri_alias}}

To join from an Infinity Connect client, click pexip:{{uri_alias}} {{app_download_
location}}

To join from a telephone, dial {{pstn_access_number}} then, when prompted, enter the
conference number {{numeric_alias}} followed by "#" {% endif %} {# if guest_pin #}
```

```
If you arrive before the meeting host, you will be placed in a virtual waiting room. You
will be automatically admitted to the meeting when the host arrives.

{% else %} {# if allow_guests #}

{% if pin %}

Your VMR is PIN-protected. How to join:

From a web browser, go to: https://{{rp_address}}/webapp/#/?conference={{numeric_
alias}}&pin={{pin}} then enter your Name and select "Connect"

From a video conferencing endpoint dial {{uri_alias}}: then enter the PIN: {{pin}}

To join from an Infinity Connect client, click pexip:{{uri_alias}}?pin={{pin}} {{app_
download_location}}

To join from a telephone, dial {{pstn_access_number}} then, when prompted, enter the
conference number {{numeric_alias}} followed by "#". When prompted, enter the PIN:
{{pin}}

{% else %} {# if pin #}

Your VMR is not protected by a PIN. This means that anyone can use the room at any time.
How to join:

From a web browser, go to: https://{{rp_address}}/webapp/#/?conference={{numeric_alias}}
then enter your Name and select "Connect"

From a video conferencing endpoint dial {{uri_alias}}

To join from an Infinity Connect client, click pexip:{{uri_alias}} {{app_download_
location}}

To join from a telephone, dial {{pstn_access_number}} then, when prompted, enter the
conference number {{numeric_alias}} followed by "#"

{% endif %} {# if pin #}

{% endif %} {# if allow_guests #}

Instructions for using your VMR can be found at https://docs.pexip.com/admin/connect_
quick.htm
```

The template above follows this basic structure:

```
<local variable definitions>
if allow_guests (VMR distinguishes between Hosts and Guests)
    <host joining instructions>
    if guest_pin
        <Guest joining instructions when there is a Guest PIN>
    else
        <Guest joining instructions when there is no Guest PIN>
    endif
else
    if pin (VMR has a Host PIN)
        <joining instructions when there is a Host PIN>
    else
        <joining instructions when there is no PIN>
    endif
endif
```

Three local variables, `pstn_access_number`, `rp_address` and `app_download_location`, are defined at the top of the template. These three lines containing the variable definitions do not appear in the generated email. The variables themselves are used at various points in the template, for example the `rp_address` variable is used as part of the URL path for connecting via a web browser. This makes it easier to update the template if the address of the reverse proxy or Conferencing Node changes.

These variable definitions are followed by an `if... else...` condition that uses the `action` variable to determine whether to generate a "new VMR" message or an "updated VMR" message.

The text included in the main body of the email is then generated according to a set of conditions that enable the relevant text to be included (such as whether to include PIN-related information or not). It starts with an outer condition of `if allow_guests` which tests whether the VMR distinguishes between Hosts and Guests, or whether the VMR has either no PIN or a Hosts-only PIN. Then within the distinguishing between Hosts and Guests section there is a nested condition that tests whether there is a Guest PIN or not. Otherwise, if the VMR does not distinguish between Hosts and Guests, there is a nested condition that tests whether the VMR has a Host PIN or no PIN at all.

Note that you can nest local variables. For example, you could define `join_webapp` as: `{%set join_webapp = "go to https://" + rp_address + "/" %}` which in this case would resolve to "go to https://rp.example.com/".

Constructing the device alias provisioning email

The **Device email subject template** and **Device email body template** fields contains the pattern for the email to be sent to the device owner when that device alias is first created (provisioned) or is updated.

The templates will typically contain a mixture of literal text and variables, and it can also contain control structures that allow you to vary the content of the email based upon certain conditions.

Supported variables

The **Device email subject template** and **Device email body template** fields support a different set of variables from the other synchronization template pattern fields (because it is based on the generated device properties, rather than the source data used to build those properties).

The following variable are available:

Variable name	Description
primary_owner_email_address	The email address of the owner of the device alias.
device_alias	The alias of the device that can be registered to Pexip Infinity.
device_description	The description of the device alias.
device_username	The username associated with the device alias. This should be used in association with the **device_password** when registering the **device_alias** to Pexip Infinity.
device_password	The password associated with the device alias.
device_tag	The device alias's service tag.
Additional generated variables	
action	Possible values are: • "created" — when the device alias has just been created • "updated" — when the existing device alias has been updated • "reminder" — when sending reminder emails.

Example device email templates

Here is an example email body template that you can use as the basis for your own emails. It is shown in plain text and in HTML format.

Basic device email template in plain text

This is a basic email template that can be used to provide the information required for registering a device.

It supplies the credentials (**device_username** and **device_password**) that must be used when registering the **device_alias** to Pexip Infinity.

If you use this example, remember to replace `confnode.example.com` with the address of your Conferencing Node.

```
Hi {{primary_owner_email_address}},

{%if action=="created" %}Your Infinity Connect client account has just been created.{%
else %}Service update: your Infinity Connect client account has been updated with new
settings. {%endif%}

This email contains the settings to use when registering your Infinity Connect client
(or your traditional hardware based video endpoint).

Alias: {{device_alias}}

Server address: confnode.example.com

User name: {{device_username}}

Password: {{device_password}}

When your device is registered, other users will be able to call you using your personal
video alias: {{device_alias}}

Instructions for registering your device can be found at https://docs.pexip.com/end_
user/guide_for_admins/registering_device_generic.htm
```

Basic device email template in HTML format

This is a repeat of the basic email template above, but illustrates how HTML formatting may be applied.

```
<html>

<p>Hi {{primary_owner_email_address}},</p>

<p>{%if action=="created" %}Your Infinity Connect client account has just been created.
{% else %}Service update: your Infinity Connect client account has been updated with new
settings. {%endif%}</p>

<p>This email contains the settings to use for your Pexip Infinity Connect client (or
your traditional hardware based video endpoint).</p>

<p>To configure your Pexip infinity client (or hardware device) you will need to enter
the following settings:</p>

<p>Alias: <b>{{device_alias}}</b></p>

<p>Server address: <b>confnode.example.com</b></p>

<p>User name: <b>{{device_username}}</b></p>

<p>Password: <b>{{device_password}}</b></p>

<p>When your device is registered, other users will be able to call you using your
personal video alias: {{device_alias}}</p>

<p>Instructions for registering your device can be found on the <a
href="https://docs.pexip.com/end_user/guide_for_admins/registering_device_generic.htm"
target="_blank">Pexip documentation website</a></p>

</html>
```

Sending reminder emails

You can manually send reminder emails for a specific VMR or device alias, or for all VMRs and devices associated with a specific sync template.

Note that no status information is shown on the **LDAP Sync Status** page when sending email reminders (as the reminder process does not resync the content from the LDAP data source). However, all related activity is recorded in the administrator log as usual.

Sending a reminder email to a VMR owner

To send a reminder email to the owner of a VMR:

1. Go to **Service Configuration > Virtual Meeting Rooms**.
2. Select the Virtual Meeting Room(s) for which you want to send reminders (select the checkbox next to the VMR name).
3. From the **Action** dropdown list, select ***Send reminder emails to VMR owners***.
4. Select **Go**.
   This will generate and send an email to the owner of each selected VMR.

To see or change the "owner" of a VMR, go into the VMR details page and show the **Advanced options**. The **Owner's email address** field is at the bottom of that section.

Sending a reminder email to a device owner

To send a reminder email to the owner of a device alias:

1. Go to **Service Configuration > Device Aliases**.
2. Select the device aliases for which you want to send reminders (select the checkbox next to the device alias name).
3. From the **Action** dropdown list, select ***Send reminder emails to device owners***.
4. Select **Go**.
   This will generate and send an email to the owner of each selected device.

To see or change the "owner" of a VMR, go into the VMR details page and review the **Owner's email address** field.

Sending reminder emails for VMRs and devices associated with a sync template

This option is useful if you have initially configured your system not to send emails — for example while you are ensuring that the VMRs and devices are being created as expected — and then want to send emails to all of those generated VMRs and devices associated with that template (as the synchronization process only automatically generates emails for an existing VMR or device if its configuration changes in some way).

To send reminder emails for all VMRs and devices associated with a specific sync template:

1. Go to **Utilities > LDAP Sync Templates**.
2. Select the sync template(s) for which you want to send reminders (select the checkbox next to the template name).
3. From the **Action** dropdown list, select ***Send VMR and device reminder emails for selected LDAP sync templates***.
4. Select **Go**.
   This will generate and send an email to the owner of each VMR and device that is associated with that LDAP sync template.

About Virtual Auditoriums

A Virtual Auditorium is a meeting space that is optimized for use by a small number of Hosts and a large number of Guests, for example:

- a lecture, where the lecturer joins as the Host and all the students join as Guests
- an internal team meeting, where the department heads join as Hosts and all the managers join as Guests.

Virtual Auditoriums differ from Virtual Meeting Rooms in that:

- Administrators can configure a Virtual Auditorium so that Hosts and Guests see different layouts. In a Virtual Meeting Room, all participants see the same layout.
- In a Virtual Auditorium, Guest participants cannot see other Guests - they will only see the Host(s). In a Virtual Meeting Room all participants can see both Hosts and Guests, in order of who spoke most recently.
- In a Virtual Auditorium, Host participants will see all other Hosts first in the video thumbnails (in order of which Host spoke most recently), followed by Guests (again, in order of who spoke most recently). In a Virtual Meeting Room, all participants can see all other participants, in order of who spoke most recently and regardless of whether they are Hosts or Guests.
- Administrators can configure a Virtual Auditorium so that when a presentation is being shown, the person showing the presentation is fixed in the main speaker position. In a Virtual Meeting Room, this option is not available; the main speaker position will always be voice-switched, showing the current speaker.

Virtual Auditoriums use PINs to protect access, and to differentiate between Host and Guest participants. For more information, see About PINs, Hosts and Guests.

You can also place a limit on the number of participants that can access a Virtual Auditorium at any one time.

To add, edit or delete a Virtual Auditorium, go to **Service Configuration > Virtual Auditoriums**. For more information, see Configuring Virtual Auditoriums.

Each Virtual Auditorium has one or more aliases associated with it. Participants access the Virtual Auditorium by dialing any one of its aliases - this will route them all to the same conference. For more information, see About aliases.

Participants can access Virtual Auditoriums from any video endpoint, or by using one of Pexip's Infinity Connect clients - for more information see Introduction to Infinity Connect.

Administrators manage Virtual Auditoriums from the Pexip Infinity Administrator interface. For more information, see Configuring Virtual Auditoriums.

There is no limit on the number of Virtual Auditoriums that can be configured on your Pexip Infinity platform. Virtual Auditoriums do not consume resources on your deployment unless they are actually being used to host a conference. Unless you manually restrict access, the number of participants who can access a particular Virtual Auditorium, and the number of Virtual Auditoriums that can be in use at the same time, are limited only by the size of your Pexip Infinity deployment.

Configuring Virtual Auditoriums

Large Pexip Infinity conferences can take place in Virtual Auditoriums. Each Virtual Auditorium has one or more aliases associated with it, and participants access the conference by dialing one of these aliases.

When creating a new Virtual Auditorium, you must provide at least one alias that will be used to access it. You have the option of configuring Host and Guest PINs for the conference.

To create a new Virtual Auditorium, go to **Service Configuration > Virtual Auditoriums** and click **Add Virtual Auditorium**.

You cannot convert an existing Virtual Meeting Room into a Virtual Auditorium. Instead, you must create a new Virtual Auditorium and reassign the existing aliases to it. For more information, see Changing from a Virtual Meeting Room to a Virtual Auditorium and vice versa.

You can manage Virtual Auditoriums using the Pexip Infinity API. For more information, see the **Pexip Infinity Management API** documentation.

To edit an existing Virtual Auditorium, or to view its details (including all the aliases associated with it), go to **Service Configuration > Virtual Auditoriums** and click on the name of the Virtual Auditorium.

Any changes you make to Virtual Auditorium configuration will be replicated to all Conferencing Nodes within 60 seconds and will be applied to any subsequent conferences in that Virtual Auditorium. If there are any conferences already in place that use the Virtual Auditorium, any attempts to join it after the configuration has been replicated may be affected by the new configuration settings. For this reason, we do not recommend changing Virtual Auditorium configuration while a conference is in progress because this will lead to an inconsistent user experience.

When adding or editing Virtual Auditoriums, the options are:

Option	Description
Name	The name used to refer to this Virtual Auditorium.
Description	A description of the Virtual Auditorium.
Creation time	This read-only field shows the date and time at which this record was first configured.

Option	Description
Host PIN	This optional field allows you to set a secure access code that must be entered by participants before they can join the conference. If **Allow Guests** is set to ***Yes***, then the **Host PIN** will apply to the conference Host(s) only. For more information, see About PINs, Hosts and Guests. • PINs must use the digits **0-9** only. • PINs may optionally end with **#**. • PINs must be between 4–20 digits long, including any **#**.
Allow Guests	***Yes***: the conference will have two types of participants, Hosts and Guests. You must configure a **Host PIN** to be used by the Hosts. You can optionally configure a **Guest PIN**; if you do not configure a **Guest PIN**, Guests can join without a PIN, but the meeting will not start until the first Host has joined. ***No***: all participants will have Host privileges. Default: ***No***.
Guest PIN	This optional field allows you to set a secure access code that must be entered by Guests before they can join the conference. For more information, see About PINs, Hosts and Guests. • Host PINs and Guest PINs must be different. • PINs must use the digits **0-9** only. • PINs may optionally end with **#**. • PINs must be between 4–20 digits long, including any **#**. • If the Host PIN ends in **#** and a Guest PIN is used, the Guest PIN must also end with **#**. • If **#** is not used, Host PINs and Guest PINs must have the same number of digits. • You cannot configure a Guest PIN unless you have already configured a Host PIN.
Host view	The maximum number of other participants that Hosts will see, and the layout used to show them. For more information, see Selecting the layout seen by participants. Default: ***Large main speaker and up to 7 other participants (1+7 layout)***.
Guest view	The maximum number of Host participants that each Guest will see, and the layout used to show them. For more information, see Selecting the layout seen by participants. Default: ***Large host speaker and up to 7 other hosts (1+7 layout)***.
Show names of speakers	If enabled, the display name or alias of each main speaker is shown. For more information, see Showing the names of speakers. Default: ***No***.
Mute all guests	When enabled, Guest participants will be muted when they first join the conference. After the conference has started, Hosts can use the Infinity Connect client to unmute Guests, either individually or as a group.
Lock presenter as main speaker	When a presentation is being shown, whether the main speaker position shows the presenter or the current speaker. For more information, see Selecting the layout seen by participants. ***Yes***: When a presentation is being shown, the main speaker position will always show the video image from the endpoint that is showing the presentation, even if others are speaking. ***No***: When a presentation is being shown the main speaker position will be voice-switched as usual. Default: ***No***.
Theme	The theme for use with this Virtual Auditorium. For more information, see Customizing images and voice prompts using themes. Default: ***<use Default theme>*** (the global default theme will be used).

Option	Description
Advanced options	
Automatically dialed participants	When a conference begins in this Virtual Auditorium, a call will be placed automatically to any participants selected here. To add an Automatically Dialed Participant that is not already on the list, select the ✚ icon to the right of the selection fields. For more information, see Automatically dialing out to a participant from a conference.
Maximum inbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being received by Pexip Infinity from each individual participant dialed in to this Virtual Auditorium. For more information see Restricting call bandwidth.
Maximum outbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being sent from Pexip Infinity to each individual participant dialed in to this Virtual Auditorium. For more information see Restricting call bandwidth.
Conference capabilities	Allows you to limit the media content of the conference. For more information, see Controlling media capability. Default: ***Main video + presentation***.
Participant limit	This optional field allows you to limit the number of participants allowed to join this Virtual Auditorium. For more information see Limiting the number of participants.
Service tag	This optional field allows you to assign a unique identifier to this service, which you can then use to track use of the service. For more information, see Tracking usage with a service tag.
Aliases	
Alias: #1	
Alias	The alias that, when received by Pexip Infinity, will cause it to route the call to this service (Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service). The alias entered here must match exactly the alias as it is received by Pexip Infinity. Wildcards and regular expressions are not supported. In most cases, the alias received by Pexip Infinity will be the same as the alias that the conference participant dialed from their endpoint, but there are some exceptions, described in About aliases. You may also want to define multiple aliases for the same service to ensure that it can be accessed by devices and protocols that enforce specific alias formats — for more information, see Using multiple aliases to access the same service.
Description	An optional description of the alias. This is useful if you have more than one alias for a service.
Add another Alias	Select this option if you want the Virtual Auditorium to be accessible by more than one alias. For more information, see Using multiple aliases to access the same service.

About the Virtual Reception IVR service

The Virtual Reception IVR service allows participants to use DTMF tones to select the Virtual Meeting Room they wish to join. It provides an alternative means to access Virtual Meeting Rooms for participants who cannot dial VMR aliases directly.

Generally, in order to access a Virtual Meeting Room, participants dial one of the Virtual Meeting Room's aliases directly from their endpoint. However, some conference participants may not be able to do this. Reasons might include:

- The participant's endpoint might not support dialing by URI - they can only dial by IP address.
- The participant might be using an audio-only PSTN telephone and can only make calls to direct dial numbers.
- Your enterprise might have a limited number of direct dial numbers that can be used for audio participants, and you do not want to allocate one per VMR.
- Your enterprise uses local, toll-free telephone numbers that audio-only users can dial to access your VMRs and you don't want to have one of these for every VMR in your enterprise.

To allow for such cases, Pexip Infinity enables you to set up a single direct dial number or IP address that participants can dial to access a single, central Virtual Reception. From here they can use the DTMF tones on their endpoint to enter the number of the specific VMR they want to join.

To implement this you must:

1. Create and configure your Virtual Reception.
2. Ensure that every Virtual Meeting Room that you want to be accessible from the Virtual Reception has at least one alias that consists of digits only. For more information, see Using multiple aliases to access the same service.
3. Provide conference participants with the combination of Virtual Reception alias followed by the Virtual Auditorium or Virtual Meeting Room number that they must dial to access the conference.

Virtual Receptions and Lync / Skype for Business clients

To allow Lync / Skype for Business clients to use Virtual Receptions, you must ensure that the target VMR is configured with at least one alias in the form of a SIP URI that is routable by the Lync / Skype for Business client. (This is in addition to the digits-only VMR alias used by the Virtual Reception.) Pexip Infinity chooses the first alias in the VMR's configuration that is a valid SIP URI; it also replaces the selected alias's domain with the **Pexip Infinity domain (for Lync / Skype for Business integration)** (from the relevant system location or global setting as appropriate, if configured).

Note that the Lync / Skype for Business client will be transferred into the destination VMR or gateway call as an audio-only participant, although they can subsequently escalate to two-way video if required (providing the **Call capability** configuration of the VMR or Call Routing Rule allows it).

Including the numeric alias of the VMR in the Virtual Reception dial string

SIP and H.323 endpoints and Lync / Skype for Business clients can optionally bypass having to enter the destination alias via DTMF tones.

They can do this by including the numeric alias of the VMR in their dial string when dialing the Virtual Reception. The dial string should be in the format: **<reception_alias>**<destination_alias>@<domain>**.

For example, if the alias of the Virtual Reception is **ivr@example.com** and the numeric alias of the target Virtual Meeting Room is **1234**, then the endpoint can dial **ivr**1234@example.com** to be transferred directly into the VMR.

Note that H.323 devices can also use the dial format **<reception_alias>#<destination_alias>@<domain>**.

Restricting or transforming the aliases entered into a Virtual Reception

To increase the security of your Virtual Reception services you may want to:

- restrict the aliases or alias patterns that can be entered into a Virtual Reception
- optionally transform the entered alias before the Virtual Reception attempts to route the call to that new destination.

You can do this when configuring your Virtual Reception by expanding the **Advanced Options** and specifying the **Destination alias regex match** and **Destination alias regex replace string** fields.

For example, you could:

- lock down the number ranges that can be reached via the Virtual Reception, e.g. by specifying a regex match of **8(.*)** to only allow calls to numbers starting with "8"
- restrict and then transform the aliases that are entered, e.g. by specifying a regex match of **(\d{4})** to only allow 4 digit extensions to be entered and then to prepend the entered alias with a prefix of "88" by entering a replace string of **88\1**
- append a domain onto any numbers entered (so that you don't need to configure so many alternative VMR aliases), e.g. by specifying a regex match of **(\d{4})** and then appending a domain by using a replace string of **\1@example.com**
- only allow access to a specified shortlist of aliases, e.g. by configuring a match string of **(1111|1112|1113)** to reject any other alias than 1111, 1112 or 1113.

Using the Virtual Reception with the Pexip Distributed Gateway

You can also use the Virtual Reception to route calls via the Pexip Distributed Gateway. This would allow you, for example, to route phone calls towards a Cisco VCS as E.164 numbers, or to join scheduled and ad hoc Lync / Skype for Business meetings.

If your environment includes a PSTN gateway or uses an ITSP (Internet telephony service provider), consider the potential for toll fraud if you have Call Routing Rules that can route calls to the PSTN gateway or ITSP, or if you allow conference participants to dial out to other participants via the PSTN gateway or ITSP. See PSTN gateways and toll fraud for more information.

Routing phone calls towards a Cisco VCS as E.164 numbers

To implement this you must:

1. Create and configure your Virtual Reception in the normal way.
2. Configure a Call Routing Rule that matches the desired E.164 pattern and then routes the call via the specified Cisco VCS (where the Cisco VCS is the SIP proxy or H.323 gatekeeper as specified in the rule).
   Ensure that the E.164 pattern does not match a Virtual Meeting Room alias (as Virtual Meeting Room routing takes precedence).

Now, if someone dials in to the Virtual Reception and then enters the E.164 number that matches the Call Routing Rule, their call will be routed via the Cisco VCS.

Joining scheduled and ad hoc Lync / Skype for Business meetings

See **Configuring Pexip Infinity as a Lync / Skype for Business gateway** for instructions on how to do this.

Configuring Virtual Reception IVRs

To create or edit a Virtual Reception IVR service:

1. Go to **Service Configuration > Virtual Receptions**.
2. To create a new Virtual Reception, select **Add Virtual Reception**, or to edit an existing Virtual Reception, click on its name.
   The following options are available:

Option	Description
Name	Enter the name you will use to refer to this Virtual Reception.
Creation time	This read-only field shows the date and time at which this record was first configured.
Description	An optional field where you can add details about the Virtual Reception.
Theme	Select the theme to apply to this Virtual Reception. For more information, see Customizing images and voice prompts using themes. If you leave this blank, the default Pexip theme will be used.
Advanced options	
Destination alias regex match	An optional regular expression used to match against the alias entered by the caller into the Virtual Reception. If the entered alias does not match the expression, the Virtual Reception will not route the call. If this field is left blank, any entered alias is permitted. For more information, see Restricting or transforming the aliases entered into a Virtual Reception.
Destination alias regex replace string	An optional regular expression used to transform the alias entered by the caller into the Virtual Reception. (Only applies if a regex match string is also configured and the entered alias matches that regex.) Leave this field blank if you do not want to change the alias entered by the caller.
Maximum inbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being received by Pexip Infinity from each individual participant dialed in to this Virtual Reception. For more information see Restricting call bandwidth.

Option	Description
Maximum outbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being sent from Pexip Infinity to each individual participant dialed in to this Virtual Reception. For more information see Restricting call bandwidth.
Conference capabilities	Allows you to limit the media content of calls that are connected via the Virtual Reception service. For more information, see Controlling media capability. Default: ***Main video + presentation***.
Service tag	This optional field allows you to assign a unique identifier to this service, which you can then use to track use of the service. For more information, see Tracking usage with a service tag.
Lync / Skype for Business server	Select a Lync / Skype for Business server only if this Virtual Reception is to act as an IVR gateway to scheduled and ad hoc Lync / Skype for Business meetings. You must then ensure that your Call Routing Rule that routes calls to your Lync / Skype for Business environment has **Match against full alias URI** selected and a **Destination alias regex match** in the style **.+@example.com.*** For more information, see **Configuring Pexip Infinity as a Lync / Skype for Business gateway**.
Lync / Skype for Business meeting lookup location	This is an optional field used in conjunction with the **Lync / Skype for Business server** setting. If specified, a Conferencing Node in this system location will perform the Lync / Skype for Business Conference ID lookup on the Lync / Skype for Business server. If a location is not specified, the IVR ingress node will perform the lookup.
Aliases	
Alias: #1	
Alias	Enter the alias that participants will dial to access the Virtual Reception.
Description	An optional description of the alias. This is useful if the Virtual Reception has more than one alias.
Add another alias	Select this option if you want the Virtual Reception to be accessible by more than one alias. For more information, see Using multiple aliases to access the same service.

3. Ensure that every Virtual Meeting Room and Virtual Auditorium that you want to be accessible from this Virtual Reception has an alias in the form of a number. To do this:
   a. Go to either **Service Configuration > Virtual Meeting Rooms** or **Service Configuration > Virtual Auditoriums**.
   b. Select the name of the Virtual Meeting Room or Virtual Auditorium.
   c. In the Aliases section at the bottom of the page, enter the number to be used to access this Virtual Meeting Room or Virtual Auditorium from the Virtual Reception.
      To allow Lync / Skype for Business clients to use Virtual Receptions, you must ensure that the target VMR is configured with at least one alias in the form of a SIP URI that is routable by the Lync / Skype for Business client. (This is in addition to the digits-only VMR alias used by the Virtual Reception.) Pexip Infinity chooses the first alias in the VMR's configuration that is a valid SIP URI; it also replaces the selected alias's domain with the **Pexip Infinity domain (for Lync / Skype for Business integration)** (from the relevant system location or global setting as appropriate, if configured).
   d. Select **Save**.

Example

In this example we want to allow participants to access any of our Virtual Meeting Rooms or Virtual Auditoriums as audio-only participants using a telephone (to allow for situations where they do not have access to a video endpoint), by making a local, toll-free telephone call to one of the numbers we have provisioned.

Step 1: Create a Virtual Reception

In order to support this, we create the following Virtual Reception:

Option	Input	Notes
Name	Toll-free Virtual Reception access	This single Virtual Reception will be accessible globally by local toll-free numbers.
Description	Allow users to dial a local telephone number and then select a VMR.	
Aliases		
Alias: #1		
Alias	15556789	The first alias is the telephone number that participants in the US will dial to access the Virtual Reception.
Description	US toll-free number	
Alias: #2		
Alias	441116789	The second alias is the telephone number that participants in the UK will dial to access the Virtual Reception.
Description	UK toll-free number	
Alias: #3		
Alias	612226789	We continue to add as many aliases as we need, one for each local telephone number that can be used.
Description	Australia toll-free number	

Now whenever participants dial any of the numbers shown above they will be taken to the same central Virtual Reception.

Step 2: Add aliases to Virtual Meeting Rooms and Virtual Auditoriums

Next, we need to ensure that our Virtual Meeting Rooms and Virtual Auditoriums are accessible from the Virtual Reception. To do this, we add a new, unique numeric alias to each Virtual Meeting Room and Virtual Reception. In this example, we want participants to be able to access Alice's Virtual Meeting Room by entering **25423** on their keypad, and the All Hands Virtual Auditorium by entering **25499**, so we edit these services as follows:

Alice's VMR

Option	Input	Notes
Name	Alice's VMR	
Description	Alice's personal meeting room	
Aliases		
Alias: #1		
Alias	meet.alice.jones@example.com	This is the existing alias for Alice's VMR.
Description	URI for Alice's VMR	
Alias: #2		
Alias	25423	We add this new alias to the existing VMR configuration.
Description	Number for Alice's VMR when accessing via the Virtual Reception.	

All Hands Virtual Auditorium

Option	Input	Notes
Name	All Hands	
Description	Virtual Auditorium for All Hands conference	
Aliases		
Alias: #1		
Alias	all-hands@example.com	This is the existing alias for the All Hands conferences.
Description	URI for All Hands conference	
Alias: #2		
Alias	25499	We add this new alias to the existing Virtual Auditorium configuration.
Description	Number for All Hands when accessing via the Virtual Reception.	

Now Alice can tell participants to join her VMR by doing any of the following:

- dialing **meet.alice.jones@example.com**
- from the US, dialing **15556789** and then entering **25423**
- from the UK, dialing **441116789** and then entering **25423**.

Likewise, participants can join an All Hands conference by doing any of the following:

- dialing **all-hands@example.com**
- from the US, dialing **15556789** and then entering **25499**
- from the UK, dialing **441116789** and then entering **25499**.

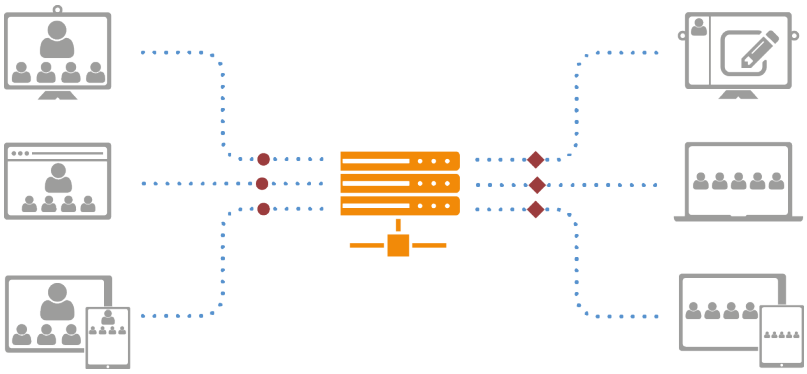
About the Pexip Distributed Gateway service

The Pexip Distributed Gateway service enables endpoints to make calls to other devices or systems that use different protocols and media formats.

Traditional hardware gateways are often expensive and are typically centralized in a single location. This means that remote endpoints making gateway calls have to route media over a WAN or over the internet which is costly and uses a lot of bandwidth.

The software-based Pexip Distributed Gateway service allows for a very cost-efficient deployment of local gateway/transcoding resources in every location. This can result in an improved user experience because of reduced latency as there is no longer a need to hairpin media back to a centralized datacenter. The other benefit is reduced WAN bandwidth usage — again due to not having to hairpin media. Reduced bandwidth usage allows an enterprise to deploy more video systems without having to upgrade the WAN infrastructure.

The Pexip Distributed Gateway service supports industry-standard protocols and emerging technologies, including H.323, SIP, Microsoft Lync / Skype for Business (MS-SIP), WebRTC and RTMP.



For example, you can use the Pexip Distributed Gateway to enable:

- users of Infinity Connect clients to place a point-to-point call to a SIP endpoint
- Lync / Skype for Business users in your

enterprise to make calls to, and receive calls from, virtually any other type of endpoint

It can also act as a gateway between Lync / Skype for Business meetings and standards-based endpoints, allowing:

- incoming calls to be routed directly into a Lync / Skype for Business meeting
- participants in a Lync / Skype for Business meeting to dial out to (i.e. invite) other non-Lync / Skype for Business participants.

How it works

Each Conferencing Node can act as a gateway service for placing calls between devices, in addition to the multi-party conferencing services offered by Pexip Infinity.

To enable devices to call other devices or systems via the Pexip Distributed Gateway, you must configure Call Routing Rules. These rules specify which calls should be interworked, for which protocols, and to where they should be routed.

Incoming calls received by Pexip Infinity are routed as follows:

1. Pexip Infinity receives an incoming call via one of its Conferencing Nodes.
2. It checks whether the destination alias belongs to a Pexip Infinity Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service; if so, it directs the call to that service.
3. If the alias does not belong to any of the above services, Pexip Infinity checks the Call Routing Rules to see if the alias matches any rules specified there for incoming calls. If so, it places an outgoing call to the destination alias according to the rule's call target settings (which protocol and call control system to use, whether to route to registered devices only, etc).

This means that if an alias matches both a Virtual Meeting Room and a Call Routing Rule, the former will always take precedence and the call will be routed to the Virtual Meeting Room. You must therefore ensure that any regular expressions used in a Call Routing Rule do not unintentionally overlap with any aliases used by Virtual Meeting Rooms, Virtual Auditoriums, Virtual Receptions or Test Call Services.

The stage where Call Routing Rules are applied in Pexip Infinity's call routing logic for incoming calls is highlighted in the following diagram:

* This information can alternatively be provided by using external or local policy

If your environment includes a PSTN gateway or uses an ITSP (Internet telephony service provider), consider the potential for toll fraud if you have Call Routing Rules that can route calls to the PSTN gateway or ITSP, or if you allow conference participants to dial out to other participants via the PSTN gateway or ITSP. See PSTN gateways and toll fraud for more information.

Note that:

- For additional security you can configure Call Routing Rules so that only registered devices are allowed to make calls via the Pexip Distributed Gateway.
- By default, the same Conferencing Node that received the incoming call is used to place the outgoing call. However, you can configure the matching rule to place the call from a Conferencing Node in a specific location. As with all calls, signaling and media may be handled by different Conferencing Nodes in that location.
- Bandwidth restrictions can be applied to gateway calls; you do this by applying the restriction to the relevant Call Routing Rule.
- If the Pexip Distributed Gateway receives DTMF signaling from an inbound call, it will generate similar DTMF on the outbound call.
- Call Routing Rules may also be applied when dialing out from a conference to a new participant (where ***Automatic*** routing has been selected). When configuring your rules, consider whether the rule is to apply to incoming calls, outgoing calls or to both incoming and outgoing calls.

Configuring Call Routing Rules

Call Routing Rules determine how calls are routed within Pexip Infinity. This includes calls received by Pexip Infinity that are to be routed via the Pexip Distributed Gateway service to their ultimate destination, and outgoing calls made from within a Pexip Infinity conference (such as when manually adding a participant to a Virtual Meeting Room).

This topic explains when Call Routing Rules are used, the main considerations when configuring your rules, how to add or modify rules, and provides an example rule.

When are Call Routing Rules used?

Call Routing Rules are always used for matching incoming calls, and may be used for matching outbound calls made from a Pexip Infinity conference.

Incoming calls

If an incoming call does not match an alias associated with a conference service (Virtual Meeting Room, Virtual Auditorium or Virtual Reception), it is deemed a Pexip Distributed Gateway call and it must match a Call Routing Rule for the call to be placed to the destination alias / target system.

Outgoing calls from a conference

Outgoing calls made from within a Pexip Infinity conference are only checked against Call Routing Rules when:

- Automatically Dialed Participants or participants added manually via the Pexip Infinity Administrator interface have their **Route this call** option set to ***Automatically***
- participants added via the Infinity Connect Web App have selected a **Protocol** of ***Automatic***

and then it must match a suitable rule for the call to be placed to the destination alias/target system.

Call routing considerations

When configuring a Call Routing Rule, the main points to consider are:

- **Rule priority**: this is the order in which the rules are considered to see if the conditions specified in the rule match the characteristics of the call that Pexip Infinity is trying to route. Rule checking stops when a match is found, even if the call that is placed as a result of that match fails for any reason.
- **Applicability to incoming or outgoing calls**: you must decide whether the rule applies only to incoming Pexip Distributed Gateway calls, or only to outgoing calls placed from within a Pexip Infinity conference (where ***Automatic*** routing has been selected), or to both incoming gateway calls and outgoing calls from conferences.

- **Restricting incoming calls to those from registered devices or specific call protocols**: for incoming calls via the Pexip Distributed Gateway, whether to limit the rule's applicability to only allow calls to be made from devices that are registered to Pexip Infinity, or to limit them to certain incoming protocols e.g. SIP only.
- **Alias matching and transforms**: the dialed alias or alias patterns that apply to this rule. A rule can be configured to apply to a specific alias e.g. alice@example.com, or to an alias pattern defined by a regex (regular expression) such as .+@example.com (any destination alias with the domain example.com). The destination alias can also, optionally, be transformed before the outgoing call is placed.
- **Call media settings**: whether to limit the media capabilities of the call, or whether to apply a theme.
- **Outgoing call placement**: every rule must define to where and how the call is routed to the destination alias. This includes defining the types of devices or systems to which the call is routed, such as limiting it to registered devices only or by specifying the SIP proxy, H.323 gatekeeper or Lync / Skype for Business server to use.

Note that any incoming call that has a destination alias that matches the alias of any Pexip Infinity Virtual Meeting Room, Virtual Auditorium or Virtual Reception is always directed automatically to that conferencing service; the call will not be routed via the Pexip Distributed Gateway and the Call Routing Rules are not applied.

The following diagram shows the call routing logic within Pexip Infinity when handling incoming call requests and when dialing out from within a conference.

* This information can alternatively be provided by using external or local policy

Note that as an alternative to using Pexip Infinity's own routing logic, you can configure Pexip Infinity to instead defer its decision making to an external policy service or to use a local policy script (see policy profiles for more information). This allows Pexip Infinity administrators to implement routing decisions based on their specific requirements.

Creating and modifying Call Routing Rules

To add, edit or delete a Call Routing Rule, go to **Service Configuration > Call Routing**.

When specifying a Call Routing Rule, the options are:

Option	Description
Name	The name used to refer to this rule.
Service tag	This optional field allows you to assign a unique identifier to this service, which you can then use to track use of the service. For more information, see Tracking usage with a service tag.
Description	An optional description of the rule.
Priority	Assign a relative priority to this rule, from 1 to 200. Rules are checked in order of priority, starting with 1 and working down the list until a match is found. Rule checking stops when a match is found, even if the call that is placed as a result of that match fails for any reason.
Use this rule for...	
Incoming gateway calls	Applies this rule to incoming calls that have not been routed to a Virtual Meeting Room or Virtual Reception, and should be routed via the Pexip Distributed Gateway service. Default: selected.
Outgoing calls from a conference	Applies this rule to outgoing calls placed from a conference service (e.g. when a participant is added to a Virtual Meeting Room) where ***Automatic*** routing has been selected. Default: not selected. Note that the same rule can be applied to both incoming and outgoing calls if required.
Calls being handled in location	Applies the rule only if the incoming call is being handled by a Conferencing Node in the selected location or the outgoing call is being initiated from the selected location. To apply the rule regardless of the location, select ***Any Location***. Default: ***Any Location***.
When matching incoming Gateway calls...	
Match incoming calls from registered devices only	Select this option if you want this rule to only apply to incoming calls from devices, videoconferencing endpoints, soft clients or Infinity Connect clients that are registered to Pexip Infinity. Note that the call must also match one of the selected protocols below. Calls placed from non-registered clients or devices, or from the Infinity Connect Web App will not be routed by this rule if it is enabled. Default: disabled.
Match Infinity Connect (WebRTC/RTMP) Match SIP Match Lync / Skype for Business (MS-SIP) Match H.323	Select the source / protocols of the incoming call to which the rule should apply. Note that matching **Lync / Skype for Business (MS-SIP)** calls does not apply if **Match incoming calls from registered devices only** is selected. Default: these options are all enabled by default.

Option	Description
Alias match and transform	
Match against full alias URI	This setting is for advanced use cases and will not normally be required. By default, Pexip Infinity matches against a parsed version of the destination alias, i.e. it ignores the URI scheme, any other parameters, and any host IP addresses. So, if the original alias received by Pexip Infinity is **sip:alice@example.com;transport=tls** for example, then by default the rule will match against **alice@example.com**. Select this option to match against the full, unparsed alias instead. You must select this option if you are using this rule to support your Lync / Skype for Business IVR gateway (where this rule is routing calls handled by a Virtual Reception into the relevant Lync / Skype for Business meeting). See **Configuring Pexip Infinity as a Lync / Skype for Business gateway** for instructions on how to do this.
Destination alias regex match	The regular expression that the destination alias (the alias that was dialed) is checked against to see if this rule applies to this call. Pexip Distributed Gateway supports case-insensitive Perl-style regular expression patterns. See Regular expression (regex) reference for information about writing regular expressions.
Destination alias regex replace string	The regular expression string used to transform the originally dialed alias (if a match was found). If you do not want to change the alias, leave this field blank.
Call media settings	
Maximum inbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being received by Pexip Infinity from each individual participant dialed in via this rule. For more information see Restricting call bandwidth.
Maximum outbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being sent from Pexip Infinity to each individual participant dialed out from this rule. For more information see Restricting call bandwidth.
Call capability	Allows you to limit the media content of the call. The participant being called will not be able to escalate beyond the selected capability. For more information, see Controlling media capability. Default: ***Main video + presentation***.
Theme	The theme to use for calls placed using this rule. For more information, see Customizing images and voice prompts using themes. Default: ***<use Default theme>*** (the global default theme will be used).
Outgoing call placement (individual fields are only displayed when appropriate)	
Call target	The device or system to which the call is routed. The options are: • ***Registered device or external system***: route the call to a matching registered device if it is currently registered, otherwise attempt to route the call via an external system such as a SIP proxy, Lync / Skype for Business server, H.323 gatekeeper or other gateway/ITSP. • ***Registered devices only***: route the call to a matching registered device only (providing it is currently registered). • ***Lync / Skype for Business meeting direct (not via Virtual Reception)***: route the call via a Lync / Skype for Business server to a Lync / Skype for Business meeting. Note that the destination alias must be transformed into a Lync / Skype for Business Conference ID. • ***Lync / Skype for Business clients, or meetings via a Virtual Reception***: route the call via a Lync / Skype for Business server either to a Lync/SfB client, or — for calls that have come via a Virtual Reception — to a Lync / Skype for Business meeting. For Lync / Skype for Business meetings via Virtual Reception routing, ensure that **Match against full alias URI** is selected and that the **Destination alias regex match** ends with **.***

Option	Description
Outgoing location	When calling an external system, this forces the outgoing call to be handled by a Conferencing Node in a specific location. When calling a Lync / Skype for Business meeting, a Conferencing Node in this location will handle the outgoing call, and — for ***Lync / Skype for Business meeting direct*** targets — perform the Conference ID lookup on the Lync / Skype for Business server. Select ***Automatic*** to allow Pexip Infinity to automatically select the Conferencing Node to use to place the outgoing call (which will usually be the node that received the call).
Protocol	The protocol used to place the outgoing call. Note that if the call is to a registered device, Pexip Infinity will instead use the protocol that the device used to make the registration. Default: ***SIP***.
SIP Proxy*	You can optionally specify the SIP Proxy to use to place an outgoing SIP call. Default: ***Use DNS***.
H.323 Gatekeeper*	You can optionally specify the H.323 Gatekeeper to use to place an outgoing H.323 call. Default: ***Use DNS***.
Lync / Skype for Business server*	When calling an external system, this is the Lync / Skype for Business server to use for outbound Lync / Skype for Business (MS-SIP) calls. When calling a Lync / Skype for Business meeting, this is the Lync / Skype for Business server to use for the Conference ID lookup and to place the call. Default: ***Use DNS*** (note that a server must be selected when the **Call target** is ***Lync / Skype for Business meeting direct (not via Virtual Reception)***).
TURN server**	You can optionally specify the TURN server to use when placing calls to ICE-enabled clients (such as Lync / Skype for Business clients and Infinity Connect WebRTC clients).
STUN server**	You can optionally specify the STUN server to use when placing calls to ICE-enabled clients (such as Lync / Skype for Business clients and Infinity Connect WebRTC clients).
Rule state	
Enable this rule	Determines if the rule is enabled or not. Any disabled rules still appear in the rules list but are ignored. Use this setting to test configuration changes, or to temporarily disable specific rules.

* If you do not select a specific **H.323 Gatekeeper**, **SIP Proxy** or **Lync server**, the Conferencing Node will attempt to use DNS to locate an appropriate system via which to route the call (rather than fall back to using the gatekeeper / proxy / server associated with the node's system location).

** If you do not select a specific **TURN server** or **STUN server**, the call will use the TURN / STUN server associated with the node's system location.

Example

In our example organization, every employee has their own video endpoint with an alias in the format **firstname.lastname@example.com**, and their own Virtual Meeting Room with an alias in the format **meet.firstname.lastname@example.com**.

In most cases, employees will use their standalone SIP or H.323 endpoints to call others within the organization, but sometimes they want to use Infinity Connect to make a point-to-point call.

In order to support this, we set up the following Call Routing Rule (unspecified settings can be left as their default values):

Option	Input	Notes
Name	Infinity Connect to SIP	

Option	Input	Notes
Description	Allow Infinity Connect (WebRTC / RTMP) users to call SIP endpoints directly	
☑ Incoming gateway calls ☐ Outgoing calls from a conference		We want this rule to match incoming gateway calls only.
Calls being handled in location	***Any location***	We want this rule to apply throughout our deployment.
☑ Match Infinity Connect (WebRTC / RTMP) ☐ Match SIP ☐ Match Lync / Skype for Business (MS-SIP) ☐ Match H.323		In this example, we only want the rule to apply to calls from Infinity Connect clients.
Destination alias regex match	.+@example.com	This regular expression will match any destination alias with the domain **example.com**.
Destination alias regex replace string	<blank>	We have left this blank because we do not want to amend the alias.
Call target	***Registered device or external system***	We want to call either a matching registered device (if it is currently registered), otherwise route the call via an external SIP system.
Outgoing location	***Automatic***	We want to place the outgoing call from the same node that received the call.
Protocol	***SIP***	The call will be interworked to SIP.
SIP Proxy	***Use DNS***	We want to use DNS to locate an appropriate SIP proxy to place the call.

This rule means that if an Infinity Connect user dials any alias with the domain **@example.com** (e.g. **alice.jones@example.com**), the call will be routed over SIP. We do not need to worry about this rule applying to VMR aliases with the same domain (e.g. **meet.alice.jones@example.com**) because Call Routing Rules are only applied to incoming calls after checking whether there are any VMRs with that alias.

Configuring the Test Call Service

Pexip Infinity provides a test loopback service that allows users to check the quality of their video and audio (i.e. that their local camera, microphone and speakers are working properly), and verifies that they can connect to a Conferencing Node.

Test Call Services are configured within Pexip Infinity in the same manner as other services such as Virtual Receptions. By default, Pexip Infinity includes a preconfigured Test Call Service (called "Test Call Service") and an associated alias of "test_call" that can be used to dial into the Test Call Service. Additional test call services and aliases can be added if required.

We recommend that you add a **test_call@<yourdomain>** alias to the preconfigured Test Call Service to cater for devices that automatically add a domain to their called alias.

After a call into a Test Call Service (e.g. to the "test_call" alias) is answered, the user receives some instructions and then the user's video and audio of themselves is played back from the Conferencing Node with a 2 second delay. The test call then automatically disconnects after approximately 20 seconds.

The instructions and timeouts can all be customized via themes, and different themes can be applied to different Test Call Services, if required.

Note that the Test Call Service does not support connections over RTMP (calls from the Infinity Connect Web App via Internet Explorer and Safari browsers).

Configuring Test Call Services and aliases

To configure Test Call Services and their associated aliases, go to **Service Configuration > Test Call Service**. The options are:

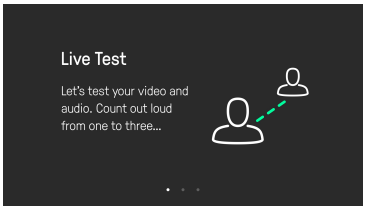
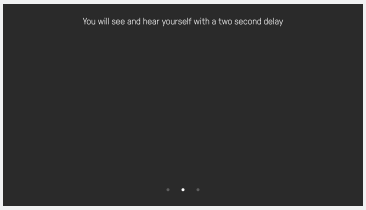
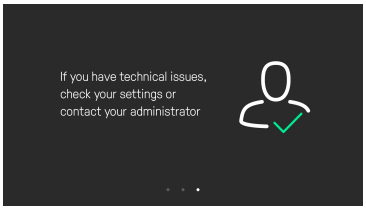
Option	Description
Name	The name used to refer to this Test Call Service.
Description	A description of the Test Call Service.
Theme	The theme for use with this Test Call Service. For more information, see Customizing images and voice prompts using themes. Default: ***<use Default theme>*** (the global default theme will be used).
Advanced options	
Maximum inbound call bandwidth (kbps)	Enter a value in this field to limit the bandwidth of media being received by Pexip Infinity from the user dialed in to this Test Call Service. For more information see Restricting call bandwidth.
Conference capabilities	Allows you to limit the media content of the conference. For more information, see Controlling media capability. Default: ***Main video + presentation***.
Service tag	This optional field allows you to assign a unique identifier to this service, which you can then use to track use of the service. For more information, see Tracking usage with a service tag.
Aliases	
Alias: #1	
Alias	The alias that, when received by Pexip Infinity, will cause it to route the call to this service (Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service). The alias entered here must match exactly the alias as it is received by Pexip Infinity. Wildcards and regular expressions are not supported. In most cases, the alias received by Pexip Infinity will be the same as the alias that the conference participant dialed from their endpoint, but there are some exceptions, described in About aliases. You may also want to define multiple aliases for the same service to ensure that it can be accessed by devices and protocols that enforce specific alias formats — for more information, see Using multiple aliases to access the same service.
Description	An optional description of the alias. This is useful if you have more than one alias for a service.
Add another Alias	Select this option if you want the Test Call Service to be accessible by more than one alias. For more information, see Using multiple aliases to access the same service.

Customizing the Test Call Service via themes

As with other Pexip Infinity services, the images that are shown, and the audio prompts that the end user hears, can all be fully customized via themes. You can also control the playback delay and the duration of the test call by configuring settings in the **themeconfig.json** theme file.

The following theme files and settings are used during a test call:

Theme file/setting	Default content	Notes
conf-test_call_ 48kHz_mono.wav	"Let's test your video and audio. Count out loud from one to three, now."	Audio file played at the start of a call to a Test Call Service.

Theme file/setting	Default content	Notes
conf-test_call_disconnect_48kHz_mono.wav	"If you have technical issues, check your settings or contact your administrator."	Audio file played at the end of a call to a Test Call Service.
test_call.jpg		Shown at the start of a call to a Test Call Service.
test_call_during_recording.jpg		Shown during a call to a Test Call Service. Note that a large, live (with a short delay) video image of the test call participant is shown on top of this JPG file during a test call.
test_call_disconnect.jpg		Shown briefly prior to automatically disconnecting the participant from a Test Call Service.
test_call_service_media_delay (in **themeconfig.json**)	2	The number of seconds that media is delayed before being looped back to the caller when using a Test Call Service.
test_call_service_disconnect_timeout (in **themeconfig.json**)	10	The number of seconds that a user can test their media before the disconnect message is played, when using a Test Call Service.

The following sequence describes the test call process and explains when each theme file and setting is used:

1. When a test call is answered, the **test_call.jpg** image is displayed and the **conf-test_call_48kHz_mono.wav** audio file is played.
2. The **test_call_during_recording.jpg** image is displayed (after the **conf-test_call_48kHz_mono.wav** audio file has finished).
3. The caller's audio and video media is replayed back to them with a **<test_call_service_media_delay>** seconds delay (2 seconds by default).
4. The media replay stops after **<test_call_service_disconnect_timeout>** seconds (10 seconds by default).
5. The **test_call_disconnect.jpg** image is displayed and the **conf-test_call_disconnect_48kHz_mono.wav** audio file is played.
6. The call automatically disconnects after a further 5 seconds (not configurable).

About aliases

Every **Virtual Meeting Room** (VMR), **Virtual Auditorium**, **Virtual Reception** and **Test Call Service** has one or more aliases associated with it.

When Pexip Infinity receives an incoming call via one of its Conferencing Nodes, it checks whether the destination alias belongs to a Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service; if so, it will direct the call to that service. If the

alias does not belong to any of the above services, Pexip Infinity will then check through the Call Routing Rules used by the Pexip Distributed Gateway to see if the alias matches any rules specified there - for more information, see Service precedence.

When Pexip Infinity receives a call to a **Virtual Meeting Room** or **Virtual Auditorium** alias, it creates a conference instance and routes the call to that conference. Any further calls received to any of the aliases belonging to the same Virtual Meeting Room or Virtual Auditorium are routed to the same conference instance, for the duration of that particular conference. If the service has more than one alias, participants can dial any one of the aliases and be routed to the same conference instance. For more information, see Using multiple aliases to access the same service.

Virtual Receptions also have one or more aliases associated with them. In these cases, when the participant dials a Virtual Reception alias they will be taken to the Virtual Reception IVR service, from which they can use a DTMF keypad to enter the number of the Virtual Meeting Room or Virtual Auditorium they wish to join. This number must correspond to a numeric-only alias of the Virtual Meeting Room or Virtual Auditorium in question. For more information, see About the Virtual Reception IVR service.

Pexip Infinity's **Test Call Service** provide a test loopback service that allows users to check the quality of their video and audio, and verifies that they can connect to a Conferencing Node. See Configuring the Test Call Service for more information.

In most cases, the alias received by Pexip Infinity will be the same as the alias that the conference participant dialed from their endpoint, but there are some exceptions, described in Search rules and alias transforms and ENUM.

For the sake of clarity, the following discussion assumes that the alias dialed by the endpoint user is the same as the alias received by Pexip Infinity.

Restrictions

An alias can include a domain but this is optional (see Domains). Aliases can be made up of:

- numbers
- letters
- dashes
- dots
- a combination of the above.

In certain circumstances an alias can also be in the form of an IP address - for more information see Dialing by IP address.

Aliases do not support wildcards or regular expressions.

Case insensitivity

The aliases that you configure on Pexip Infinity are not case-sensitive, and Pexip Infinity treats all incoming aliases as not case-sensitive.

For example, this means that:

- a user who dials `meet.alice` will match against a VMR with an alias of `Meet.Alice`
- a user who dials `Meet.Alice` will match against a VMR with an alias of `meet.alice`

Ignoring IP addresses

Pexip Infinity ignores any IP address or IP address and port combination appended to an alias. This is because some SIP endpoints automatically add the IP address of their proxy to the URI that is dialed by the user.

For example, a VMR with a single alias of `meet.alice` will be matched when an endpoint dials any of:

- `meet.alice`
- `meet.alice@<IPaddress>`
- `meet.alice@<IPaddress>:<port>`

Dialing by IP address

We do not recommend dialing by IP address. However, to support endpoints that can only dial by IP address, you can give a Virtual Reception, Virtual Meeting Room or Virtual Auditorium an alias that is the same as the IP address of one of your Conferencing

Nodes.

In this case, when a user dials the IP address from that endpoint, the call will be routed directly to the Conferencing Node with that IP address. Pexip Infinity will then perform its standard behavior and check to see if the destination alias of the call (which in this scenario will be the dialed IP address) matches any of its configured aliases; if so, the call will be routed to the service (Virtual Reception, Virtual Meeting Room etc.) that is associated with that alias.

If other endpoints, such as Infinity Connect clients, dial the IP address alias, they will also be connected to the appropriate service (although the Conferencing Node they connect to will be determined via other means).

In most cases you would use this feature to assign a Conferencing Node IP address as an alias for a Virtual Reception, so that users could then select which Virtual Meeting Room or Virtual Auditorium they wish to join. For more information, see About the Virtual Reception IVR service.

Ignoring protocol prefixes

Pexip Infinity ignores relevant URI schemes included in an alias, as in the case where a SIP endpoint automatically prefixes the URI dialed by the user with `sip:`. The ignored prefixes are:

- `sip:`
- `sips:`
- `h323:`

This is because the protocol used for the call does not matter to Pexip Infinity.

For example, a VMR with a single alias of `meet.alice` will be matched when an endpoint dials any of:

- `sip:meet.alice@IPaddress`
- `sips:meet.alice@IPaddress`
- `h323:meet.alice`

Search rules and alias transforms

Some call control systems (for example the Cisco VCS) support the use of search rules and alias transforms. In these cases, the alias that was dialed by the endpoint user and received by the call control system may be changed by the call control system in some way before it is passed on to Pexip Infinity. Often this is done in larger deployments to support complex dial plans.

For example, the call control system might have a rule that replaces the domain `example.com` with `example.net`. This means that when a user dials `meet.alice@example.com`, the call is routed to `meet.alice@example.net`. Therefore it is this latter alias that must be configured on Pexip Infinity in order to match it with a VMR.

ENUM

The ENUM system allows users to dial an E.164 number (for example a telephone number) which is then mapped using DNS to a SIP URI. For more information, see **RFC 3761**.

For example, your dial plan could be set up so that when a user dials `555 0123`, the call is routed via ENUM to `meet.alice`.

If your dial plan uses ENUM, the resulting SIP URIs must be included as aliases on Pexip Infinity in order to match them with a VMR.

Domains

Domains are optional in aliases. However, when a call is received to an alias in the form of a URI that includes a domain, the domain is not ignored.

For example, if a VMR is configured with an alias of `meet.alice`, users can dial `meet.alice` or `meet.alice@<IPaddress>` to access the meeting room. However, if they dial `meet.alice@example.com` they will not be able to access the VMR because the VMR alias does not include the domain.

When a SIP endpoint user dials an alias that does not include a domain (for example `meet.alice`), the SIP endpoint will automatically add its own domain to the alias (making it for example `meet.alice@example.com`). So even if the VMR is configured with an alias of `meet.alice` and this alias is dialed by a participant from a SIP endpoint, the participant will not be able to join the conference. (H.323 endpoints do not add domains.)

In the absence of a call control system to strip the domain part of the alias, you could instead add to the VMR a second alias of `meet.alice@example.com` so that participants can dial `meet.alice` from either a SIP or H.323 endpoint and access the same conference.

Using multiple aliases to access the same service

You may want conference participants to be able to dial different numbers to access the same Virtual Reception, or to allow different conference participants to dial different aliases but still end up in the same VMR.

Thus, depending on your deployment environment and the types of devices and protocols used to dial into your services, you may want to configure a variety of alias formats to allow access to those services, including:

- **name@domain** (for full URI dialing such as from SIP and Lync / Skype for Business clients)
- **name** (for systems that strip off or do not require a domain)
- **numeric/E.164** (for endpoints that do not support URI dialing, or can only dial via IP address, or for audio-only callers via PSTN/ISDN gateways that need to be routed via a Virtual Reception).

It is often good practice to create a URI and a numeric alias as a minimum. If numeric/E.164 formatted aliases are used as your primary dial plan, you could also reuse the same number in a number@domain URI-style alias. Some example alias dial plans are given below.

You can add any number of aliases to a single Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service. (Note that the order in which aliases appear in the list of aliases is relevant when Automatically dialing out to a participant from a conference.)

Adding additional aliases

To add an additional alias to a service while you are creating it, select **Add another Alias**.

To add another alias to an existing service:

1. Go to:
   - **Service Configuration > Virtual Meeting Rooms**,
   - **Service Configuration > Virtual Auditoriums**,
   - **Service Configuration > Virtual Receptions**, or
   - **Service Configuration > Test Call Service**.
2. Select the name of the service you wish to add the alias to.
3. In the **Aliases** section at the bottom of the page, enter the new alias in the empty **Alias** field.
4. Select **Save**.

Examples

- If some of your endpoints do not support URI dialing, you could set up a VMR for Alice with one alias of **meet.alice.jones** and another alias of **555 25423**. This would give conference participants two different methods of accessing the same conference.
- If your deployment includes a Virtual Reception, every Virtual Meeting Room and Virtual Auditorium that you want to be accessible from the Virtual Reception will need to have a numeric-only alias (so that participants can enter the number using DTMF), in addition to any aliases in the form of a URI. So to continue with our example, you could add a third alias to Alice's VMR of **25423**, which is the number that participants would enter from the Virtual Reception. For more information, see About the Virtual Reception IVR service.
- If you use an ISDN gateway to support telephone participants, you could set up Alice's VMR with aliases of **meet.alice.jones** and an appropriate E.164 number.
- If you do not have or do not wish to use a call control system to transform aliases, you could set up Alice's VMR with aliases of **meet.alice.jones** and **meet.alice.jones@example.com** to support internal and external dialing, and dialing from both SIP and H.323 endpoints.

Creating and editing aliases

Aliases are the strings that users dial to access Pexip Infinity services - Virtual Meeting Rooms, Virtual Auditoriums, Virtual Receptions and Test Call Services. Each alias is associated with a single service, but a service can have more than one alias - for more information, see Using multiple aliases to access the same service.

To add a single alias to an existing service, go to **Service Configuration > Aliases** and click **Add Alias**. This will take you to the **Add Alias** page.

You can also add a new alias while creating or editing a service. For further information, see Configuring Virtual Meeting Rooms, Configuring Virtual Auditoriums, Configuring Virtual Reception IVRs and Configuring the Test Call Service.

When adding or editing aliases from the **Add Alias** page, the options are:

Option	Description
Service	Select the name of the Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service that will be accessed when participants dial this alias.
Alias	The alias that, when received by Pexip Infinity, will cause it to route the call to this service (Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service). The alias entered here must match exactly the alias as it is received by Pexip Infinity. Wildcards and regular expressions are not supported. In most cases, the alias received by Pexip Infinity will be the same as the alias that the conference participant dialed from their endpoint, but there are some exceptions, described in About aliases. You may also want to define multiple aliases for the same service to ensure that it can be accessed by devices and protocols that enforce specific alias formats — for more information, see Using multiple aliases to access the same service.
Description	An optional description of the alias. This is useful if you have more than one alias for a service.

Registering devices to Pexip Infinity

Pexip Infinity acts as a SIP registrar and H.323 gatekeeper. In addition to SIP and H.323 endpoints, Infinity Connect desktop clients and Infinity Connect Mobile clients for Android can also register to Pexip Infinity Conferencing Nodes. This allows Pexip Infinity to route outbound calls to those registered devices without having to go via a SIP proxy or H.323 gatekeeper, or rely on DNS.

Devices can only register to Pexip Infinity with a permitted alias and by supplying valid credentials (if authentication is required). Allowed aliases and their associated credentials can be configured manually, or they can be bulk provisioned from directory information contained in a Windows Active Directory LDAP server, or any other LDAP-accessible database (see Provisioning VMRs and devices from Active Directory via LDAP).

Note that devices do not need to be registered in order to make inbound calls to Pexip Infinity. However, you can configure Call Routing Rules so that only registered devices are allowed to make calls via the Pexip Distributed Gateway.

For end-user instructions on how to register a device, see Registering your device to receive calls.

Configuration summary for enabling registrations

To configure Pexip Infinity to accept registrations and route calls to registered devices you must:

1. Ensure that the Pexip Infinity's registrar service is enabled (**Service Configuration > Registrar**).
2. Configure the aliases that devices are allowed to register to Pexip Infinity (**Service Configuration > Device Aliases**).
3. Configure appropriate Call Routing Rules (**Service Configuration > Call Routing**). Rules are required in all cases except for when using ***Manual*** routing to add a participant to a conference.

Each step is described in more detail below, after the explanation of how Pexip Infinity manages registered devices.

How it works

Registering devices

The registrar service on Pexip Infinity is enabled by default. However, a device can only register to Pexip Infinity if the alias it wants to register has been added to Pexip Infinity's list of allowed device aliases (**Service Configuration > Device Aliases**).

Device registrations can optionally also be subject to authentication. To enforce authentication, username and password credentials must be included when the alias is added to Pexip Infinity's list of allowed device aliases. When credentials have been included, the device's registration request will be challenged and the device will only be allowed to register with that alias if it provides matching credentials. Pexip Infinity supports Digest authentication only.

A device can register to any Conferencing Node. Pexip Infinity does not apply any form of registration load balancing or failover. Multiple devices can register with the same alias.

- To register a **SIP device**, use the IP address or FQDN of a local Conferencing Node as the SIP proxy.
- To register an **H.323 device**, use the IP address or FQDN of a local Conferencing Node as the H.323 gatekeeper.

  All of the device aliases presented in the H.323 registration request must be in the list of allowed device aliases. If any alias is not present, none of the aliases will be allowed to register. Additionally, if device authentication is being used, all of the aliases presented in the request must use the same credentials.
- If your endpoint supports SIP and H.323, you should register it to Pexip Infinity over just one protocol, either SIP or H.323, but not both.
- To register an **Infinity Connect client**, ensure that either the IP address or FQDN of a local Conferencing Node, or the IP address or FQDN of the reverse proxy, is used as the registration server address (see About Infinity Connect client settings for more information).

  Infinity Connect clients register over the WebRTC protocol.

  When an Infinity Connect client is registered, as well as being able to receive calls, the user can filter and lookup the contact details (phone book / directory) of other devices or VMRs that are set up on the Pexip Infinity platform, making it easier to call those devices or VMRs. For more information, see Registering devices to Pexip Infinity.

After the device has registered it must periodically refresh that registration.

Calling registered devices

Whenever Pexip Infinity needs to place an outbound call (such as when adding a participant to a conference, or when making a call via the Pexip Distributed Gateway), it will first look through the set of devices currently registered to any of the Conferencing Nodes on the Pexip Infinity platform. If the alias it is trying to call is currently registered, then Pexip Infinity will place the call to that registered device.

When calling a registered device:

- the Conferencing Node to which the device is registered will place the call to the device (media may flow through other nodes, as per the platform's standard distributed conferencing behavior)
- the outbound call is always placed to the registered device over the same protocol that it used to make the registration
- if multiple devices are registered with the same alias, Pexip Infinity will place a call to each device, i.e. it will fork the call.

Requirement for Call Routing Rules

Note that, in most cases, you must also configure a suitable Call Routing Rule that instructs Pexip Infinity to place the outbound call (either to the registered device or to any other non-registered device or system). A Call Routing Rule **is** required when:

- making a call to another device or system via the Pexip Distributed Gateway
- using ***Automatic*** routing when adding a participant to a conference.

A Call Routing Rule **is not** required when:

- using ***Manual*** routing when adding a participant to a conference.

When configuring rules you must define:

- whether the rule applies only to incoming Pexip Distributed Gateway calls, or only to outgoing calls placed from within a Pexip Infinity conference (where ***Automatic*** routing has been selected), or to both incoming gateway calls and outgoing calls from

conferences
- the destination aliases to be matched — this needs to match the pattern of the registered device aliases
- whether to route the call to ***Registered devices only*** (in which case the call will fail if the device is not currently registered), or to **Registered device or external system** which will route the call to a matching registered device if it is currently registered, otherwise it will attempt to route the call via an external system.

Avoiding call looping

Call Routing Rules that are intended to route calls to devices registered to Pexip Infinity (e.g. Infinity Connect clients) could, depending upon DNS configuration, result in call looping if the destination device alias is not currently registered.

This can occur if the Call Routing Rule handling those calls has its **Call target** set to ***Registered device or external system***. In this case, if the alias is not currently registered, Pexip Infinity will attempt to place the call via DNS or a call control system. This could result in the call being routed back to Pexip Infinity which would then match the same Call Routing Rule and result in a loop.

To avoid this, we recommend that you configure your Call Routing Rules that match the pattern of registered device aliases to use a **Call target** of ***Registered devices only***. This ensures that the call will fail cleanly without looping if the device is not currently registered.

To avoid this, we recommend that you configure a **Call target** of ***Registered devices only*** for your Call Routing Rules that match registered device aliases. This ensures that the call will fail cleanly without looping if the device is not currently registered.

Note that in earlier versions of Pexip Infinity before the availability of the ***Registered devices only*** option, we previously recommended configuring a SIP proxy with an address of **nowhere.invalid**. This workaround will still avoid the call looping issue, but if you have any existing rules that use this method, we recommend modifying them to use a **Call target** of ***Registered devices only*** instead.

Call routing logic

The stage in Pexip Infinity's call routing logic where it checks if a device is registered is highlighted in the following diagram:

* This information can alternatively be provided by using external or local policy

Note that you can configure Call Routing Rules so that only registered devices are allowed to make calls via the Pexip Distributed Gateway.

Configuring the registrar service

Devices can only register to Pexip Infinity if the registrar service is enabled.

To configure Pexip Infinity's registrar service, go to **Service Configuration > Registrar**. The options are:

Option	Description
Enable registrar	Controls whether devices can register to Pexip Infinity. Devices must register with a permitted alias (configured via **Service Configuration > Device Aliases**). The registrar service is enabled by default.
Registration refresh (general)	
Registration refresh strategy	Defines which strategy to use when calculating the expiry time of a SIP or H.323 registration: • ***Adaptive***: Pexip Infinity automatically adjusts the refresh interval depending on the number of current registrations on the Conferencing Node handling the registration request, in order to spread the load of registration refreshes. As the number of devices registered to a Conferencing Node increases, the refresh interval calculated by Pexip Infinity will also typically increase, although it will always be within the bounds of the configured minimum and maximum refresh intervals. • ***Basic***: Pexip Infinity simply uses the configured minimum and maximum settings, along with the requested value, to determine the refresh interval. Default: Adaptive.
Minimum refresh interval (Adaptive strategy)	The minimum interval in seconds before a device's registration must be refreshed, when using the ***Adaptive*** strategy. Default: 60 seconds.
Maximum refresh interval (Adaptive strategy)	The maximum interval in seconds before a device's registration must be refreshed, when using the ***Adaptive*** strategy. Default: 3600 seconds.
Minimum refresh interval (Basic strategy)	The minimum interval in seconds before a device's registration must be refreshed, when using the ***Basic*** strategy. Default: 60 seconds.
Maximum refresh interval (Basic strategy)	The maximum interval in seconds before a device's registration must be refreshed, when using the ***Basic*** strategy. Default: 300 seconds.
Registration refresh intervals for SIP endpoints behind NATs	
Minimum refresh interval (when behind NAT)	The minimum interval in seconds before a device's registration must be refreshed, for a SIP endpoint behind a NAT. Default: 60 seconds.
Maximum refresh interval (when behind NAT)	The maximum interval in seconds before a device's registration must be refreshed, for a SIP endpoint behind a NAT. The refresh interval for SIP endpoints behind a NAT typically has to be shorter than the interval for other endpoints. This is to help keep the NAT pinhole alive. Default: 90 seconds.

Note that:

- In all circumstances, requests for a value lower than the relevant **Minimum refresh delay** will result in the registration being rejected.
- The registration refresh strategies apply only to SIP and H.323 endpoints. Infinity Connect clients have a fixed refresh interval of 120 seconds.

Specifying the aliases that devices are allowed to register with

Devices can only register to Pexip Infinity with a permitted alias.

You can bulk provision the device aliases from directory information contained in a Windows Active Directory LDAP server, or any other LDAP-accessible database. For more information, see Provisioning VMRs and devices from Active Directory via LDAP.

You can also import device aliases using a CSV file. For more information, see Bulk import/export of device aliases.

To manually configure the aliases that devices are allowed to register to Pexip Infinity, go to **Service Configuration > Device Aliases**. The options are:

Option	Description
Device alias	The alias URI with which the device must register to Pexip Infinity. The alias must be an exact match; regular expressions are not supported.
Creation time	This read-only field shows the date and time at which this record was first configured.
Service tag	This optional field allows you to assign a unique identifier to this service, which you can then use to track use of the service. For more information, see Tracking usage with a service tag.
Description	An optional description of the device.
Username and Password	The username and password credentials with which to authenticate the device. The credentials are optional and the device is not challenged if no credentials are entered. The username and password are case-sensitive.
Device origin	The name of the LDAP sync template used to create this device alias; if the device was created by manual input or via the API this will be blank. This field is read-only.
Owner's email address	The email address of the owner of the device alias. Device provisioning messages associated with this alias will be sent to this address.

Directory (phone book) of devices and VMRs for registered Infinity Connect clients

Pexip Infinity provides a directory search facility to all Infinity Connect clients that are registered to a Conferencing Node.

Upon request from the Infinity Connect client (typically when the user performs a search on their contact list), Pexip Infinity will return a list of device and conference aliases that match the search string entered by the user.

Pexip Infinity checks the supplied search string to see if it is contained within the alias or description of any of its configured services (Virtual Meeting Rooms, Virtual Auditoriums or Virtual Receptions), or within the device aliases that are allowed to register to Pexip Infinity (it does not matter if that device alias is currently registered or not, and no presence information is supplied).

Pexip Infinity returns up to 5 service aliases and up to 5 device aliases.

The directory service can be enabled or disabled via the **Enable directory** global setting (**Platform Configuration > Global Settings**). It is enabled by default.

Feature extensions via the external policy API

If you make use of the external policy API, the external system can be used to:

- determine whether a device alias is allowed to register to a Conferencing Node.
- obtain extended directory information — this can replace or be used in addition to the list of local device and VMR aliases supplied to registered Infinity Connect clients.

See **Using external and local policy to control Pexip Infinity behavior** for more information.

Troubleshooting and logging

To see a list of all device aliases that are currently registered to the Pexip Infinity platform, go to **Status > Registrations**.

Successful registration requests and expired registrations are recorded in the administrator log.

Failed registration requests (due to, for example, an unknown device alias or authorization failure) and refreshed registrations are recorded in the support log only.

If a call fails to be placed to a registered device, check that an appropriate Call Routing Rule has been configured that has suitable destination alias matching strings for the registered alias (and ensure that call looping cannot occur).

Customizing the Infinity Connect Web App

The Infinity Connect Web App is available as part of all Pexip Infinity deployments. It provides a WebRTC or Flash-based interface to Pexip Infinity services.

The standard Infinity Connect Web App uses Pexip logos and icons and has an orange color scheme. However, you can customize all text, color, and image elements to provide a branded experience. To do this you must create and then upload a branding package to the Management Node.

Branding customizations that are applied via the Management Node will persist over upgrades to subsequent versions of Pexip Infinity software (although you may need to adapt the customization to cater for any new features when upgrading to a new major release).

The procedures described here apply a generic customization for all Web App users. If you have specific customization requirements, such as hosting multiple different branding customizations under different URLs on external web servers or reverse proxies, see Advanced Infinity Connect Web App customization.

The following instructions describe how to create and upload, edit and remove a branding package.

Creating and uploading a branding package

You must first create a branding package before you can upload it to the Management Node.

There are two main ways to create a branding package for the Infinity Connect Web App:

- via the Pexip branding portal (https://branding.pexip.com)
- manual configuration of the default branding files after downloading them from the Management Node

Note that both methods use the same set of configuration files — you can use the branding portal to easily apply your basic customization requirements and then make further manual amendments to the configuration files if necessary.

If you also use the Infinity Connect desktop client, you can use the same set of customized files in both the Infinity Connect Web App and the Infinity Connect desktop client. See Customizing the Infinity Connect desktop client for more information.

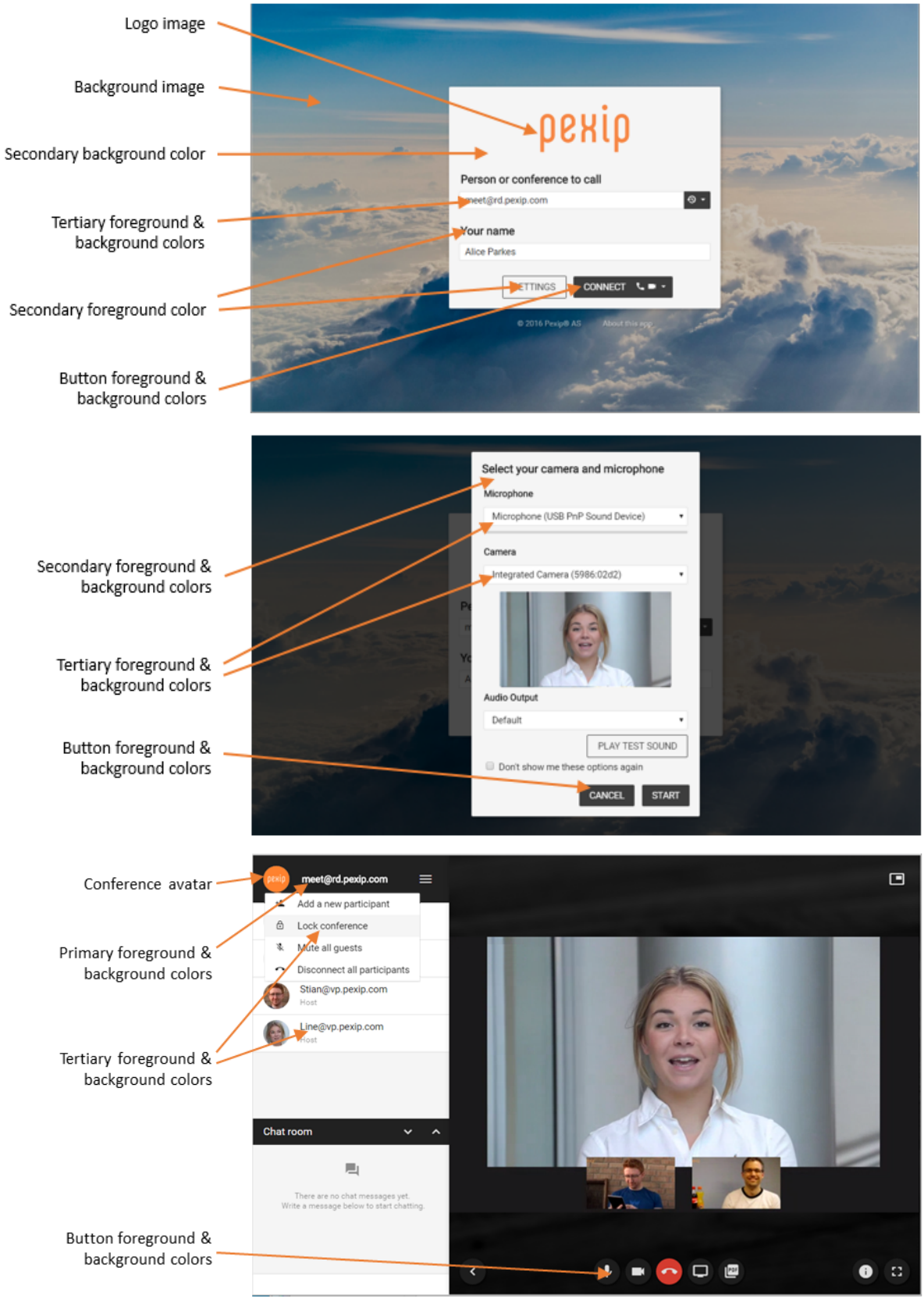
Creating a branding package via the Pexip branding portal

You can use the Pexip branding portal to customize your Web App. This web-based tool guides you through the selection of your image files and colors without having to edit individual CSS files etc., and then generates the customized branding package for you. The portal also lets you go back and modify existing branding packages and share your branding with colleagues from your organization.

To use the Pexip branding portal to generate your branding package:

1. Go to the Pexip branding portal (https://branding.pexip.com/) and sign in.
   First time users need to register before they can use the portal.
2. From here you can choose to add a new customization (branding package), or edit an existing customization that you or somebody else in your organization has previously created.
3. Configure your customization as required, selecting the relevant image files, colors and languages.
   Note that you can use the **Preview** option to see how the Web App home page would look with your branding applied.

The following screenshots show the default branding and indicate which elements of the Web App are controlled by the various image and color options presented in the branding portal.



4. When you have finished configuring your branding, select **Download** and choose the Pexip MCU software version as appropriate for your installation.
   This creates and downloads a **<customization _name>.zip** file containing your customizations.
5. Upload the branding package to your Management Node:
   a. Go to **Service Configuration > Web App Customization**.
   b. In the **Upload Web App branding** section, select **Choose File** and select the ZIP file containing your customizations.
   c. Select **Upload branding**.
      The branding package will be uploaded.
   Allow a minute or so for the new branding to be replicated out to all Conferencing Nodes.

Manually configuring the branding files

Manual configuration is useful if you have very specific modifications that you want to apply to the branding files, or if you want to add a language that is not available on the branding portal. Note that manual configuration requires knowledge of core web design technologies such as HTML, JavaScript and CSS.

To manually configure the branding files:

1. Download the default Web App branding files from the Management Node:
   a. Go to **Service Configuration > Web App Customization**.
   b. Select **Download** (next to the **Download default branding** label). This downloads a **branding_default.zip** file to your local file system.
   Note that if you have existing branding files uploaded, you can choose to download those (as **branding_custom.zip**) instead of the default files.
2. Unpack the downloaded file and apply your modifications to the relevant files.
   The contents of the branding files and how to modify them is fully described in Manually customizing the application files.
3. Repackage your branding files into a single .ZIP file (**<name>.zip**).
   The .ZIP file does not have to contain the complete set of branding files. You can upload a subset of the branding files (for example, if you have only changed some of the images or aspects of the CSS), but you must retain the original file/folder structure in the rebuilt zip file.
   Note that if you change the **background.jpg** or **logo.png** graphics files you must also include a **brand.css** file that at least includes the references (**brand-logo** and **brand-background** classes) to those customized images.
4. Upload the branding package to your Management Node:
   a. Go to **Service Configuration > Web App Customization**.
   b. In the **Upload Web App branding** section, select **Choose File** and select the ZIP file containing your customizations.
   c. Select **Upload branding**.
      The branding package will be uploaded.
   Allow a minute or so for the new branding to be replicated out to all Conferencing Nodes.

You can now test the branding by dialing in to one of your Pexip Infinity services via the Infinity Connect Web App.

Editing an existing branding package

You can modify an existing branding package by either returning to Pexip branding portal or manually editing the branding files that were uploaded previously to the Management Node.

Note that when you upload a new branding package to the Management Node, all of the previously uploaded branding files are deleted and replaced with the new set of files.

Using the branding portal

If you initially created your branding package via the Pexip branding portal, you can return to the portal and change those files:

1. Go to the Pexip branding portal (https://branding.pexip.com/) and sign in.
2. Select the customization (branding package) you want to change.
3. Make your changes, previewing them if necessary, and then download a new ZIP file.
4. On the Management Node, go to **Service Configuration > Web App Customization** and upload your new ZIP file.

Allow a minute or so for the new branding to be replicated out to all Conferencing Nodes.

Manually changing your existing branding on the Management Node

You can manually edit the existing branding files that have been uploaded to the Management Node (even if those files were originally created via the Pexip branding portal):

1. On the Management Node, go to **Service Configuration > Web App Customization**.
2. Download the existing branding files (select the **Download** option next to the **Download current branding** label).
3. Unpack the downloaded file and apply your modifications to the relevant files.
   The contents of the branding files and how to modify them is fully described in Manually customizing the application files.
4. Repackage your modified branding files into a single .ZIP file.
5. Upload the new ZIP file back onto the Management Node (**Service Configuration > Web App Customization** then **Choose File** followed by **Upload branding**).

Allow a minute or so for the new branding to be replicated out to all Conferencing Nodes.

Removing a branding package (revert to default branding)

If you want to revert to the default Infinity Connect Web App branding, you need to remove your customized branding from the Management Node. To do this:

1. On the Management Node, go to **Service Configuration > Web App Customization**.
2. From the bottom-right corner of the page, select **Remove branding** and confirm.

Allow a minute or so for the customized branding to be removed from all Conferencing Nodes and for them to revert to the default branding.

Managing conferences

About PINs, Hosts and Guests

For added security, you can set up your Pexip Infinity Virtual Meeting Rooms and Virtual Auditoriums with PIN numbers. You can use the same PIN for all participants, or you can use PINs to differentiate between Hosts and Guests. When a participant dials the alias of a Virtual Meeting Room or Virtual Auditorium that has a PIN, they are presented with an Interactive Voice Response (IVR) screen where they are asked to enter the PIN number. They must enter the correct PIN before they can join the conference.

This topic covers:

- Using the same PIN for all participants
- Using PINs to differentiate between Hosts and Guests
- Using # at the end of a PIN
- Changing a participant's role from Guest to Host (and vice versa)
- How to combine Host PINs and Guest PINs for differing levels of security

Using the same PIN for all participants

All participants in a conference that uses a single PIN will have the same Host privileges. (To create a PIN-protected conference where participants have different levels of privileges, see Using PINs to differentiate between Hosts and Guests.)

Participants have 3 attempts to enter a correct PIN. After the third unsuccessful attempt, they will hear a message advising them that the PIN is invalid, and their call will be disconnected.

All PIN entry attempts, successful and unsuccessful, are logged in the administrator log.

To set a PIN number on a Virtual Meeting Room:

1. Go to **Service Configuration > Virtual Meeting Rooms**.
2. Select the Virtual Meeting Room.
3. In the **Host PIN** field, enter the PIN number to be used.
4. In the **Allow Guests** drop-down list, select ***No***.

Note that:

- PINs must use the digits **0-9** only.
- PINs may optionally end with **#**.
- PINs must be between 4–20 digits long, including any **#**.

Using PINs to differentiate between Hosts and Guests

You can set up your Pexip Infinity Virtual Meeting Rooms and Virtual Auditoriums so that they are "Hosted". Hosted Virtual Meeting Rooms and Virtual Auditoriums have two different types of participants: **Hosts** and **Guests**. Hosts and Guests both dial the same alias in order to join the conference, however:

- Only those participants who enter the **Host PIN** will have privileges to control the conference; everyone else is a Guest and can only view the conference.
- If a conference is locked, Hosts will still be able to join but Guests will be held at a waiting screen.
- For conferences that use Virtual Auditoriums, the administrator can select different layouts for the Host view and Guest view. For more information, see Selecting the layout seen by participants.

Guest privileges

- If any Guests join the conference before a Host has arrived, they will be shown a holding image and hear a message advising them that they are waiting for the conference Host.

- A minute or so after the last Host has left, any remaining Guests will be automatically disconnected. During this time, Virtual Auditorium Guests will be able to see other Guests in the selected layout. (Note that administrators can change the length of time before Guests will be disconnected by going to **Platform Configuration > Global Settings > Guests-only Timeout**.)

In addition, if the conference is held in a Virtual Auditorium:

- Guest will only be able to see the Hosts; they will not be able to see other Guests. For more information, see Selecting the layout seen by participants.

Host privileges

- Normally, Guests cannot join a conference until the first Host has joined. This does not apply if the Host is control-only, so control-only Hosts can elect to start the conference manually, thus allowing Guests to join.
- If the first participant to join the conference is a Host, they will be shown a holding image until another Host or Guest joins the conference.
- If the conference has a Host PIN, participants who enter the Host PIN will be able to join a conference even if it is locked.
- The presence of a Host in a conference may prevent that conference from being automatically terminated.
- A minute or so after the last Host has left, any remaining Guests will be automatically disconnected. (Note that administrators can change the length of time before Guests will be disconnected by going to **Platform Configuration > Global Settings > Guests-only Timeout**.)
- Hosts can use Infinity Connect clients to:
  - disconnect participants from the conference
  - add participants to the conference
  - mute and unmute an individual participant, or all Guest participants simultaneously
  - lock and unlock a conference, and to allow individual participants in to a locked conference
  - change the role of other participants in the conference
  - control another participant's camera
  - send DTMF tones to other participants in the conference.

If the conference is taking place in a Virtual Auditorium:

- The Pexip Infinity administrator may configure the Virtual Auditorium so that Hosts see other participants in a different layout to that seen by Guests. For more information, see Selecting the layout seen by participants.

Host and Guest PINs

You can create two types of Hosted conference:

- the Host must enter a PIN but Guests do not, or
- both Hosts and Guests must enter a PIN.

Participants have 3 attempts to enter a correct PIN. After the third unsuccessful attempt, they will hear a message advising them that the PIN is invalid, and their call will be disconnected.

All PIN entry attempts, successful and unsuccessful, are logged in the administrator log.

Creating a Hosted Virtual Meeting Room or Virtual Auditorium

To create a Hosted Virtual Meeting Room or Virtual Auditorium:

1. Go to either:
   - **Service Configuration > Virtual Meeting Rooms**
   - **Service Configuration > Virtual Auditoriums**.
2. Select the Virtual Meeting Room or Virtual Auditorium.
3. In the **Host PIN** field, enter the PIN number to be used by the Hosts.
4. In the **Allow Guests** drop-down list, select ***Yes***.
5. If you want to require Guests to enter a PIN, in the **Guest PIN** field, enter the PIN number to be used.
   If you leave this field blank, anyone who dials the alias of the Virtual Meeting Room or Virtual Auditorium will be able to access

the conference as a Guest. However, the conference will not start until the first Host has joined and in the meantime all Guests will continue to be shown the holding screen.

Note that:

- Host PINs and Guest PINs must be different.
- PINs must use the digits **0-9** only.
- PINs may optionally end with **#**.
- PINs must be between 4–20 digits long, including any **#**.
- If the Host PIN ends in **#** and a Guest PIN is used, the Guest PIN must also end with **#**.
- If **#** is not used, Host PINs and Guest PINs must have the same number of digits.
- You cannot configure a Guest PIN unless you have already configured a Host PIN.

Using # at the end of a PIN

PINs can optionally end with a **#** (known either as the **pound** or **hash** character). Using a terminal **#**:

- Allows administrators to set different length PINs for Hosts and Guests.
- Allows users who have accidentally entered too many or too few digits (particularly those using audio-only, who have no on-screen indication of the number of digits required) to return to the start of the PIN entry process by entering **#**.

The Administrator interface will prevent you from entering a Host PIN that ends with # and a Guest PIN that does not (or vice versa).

For conferences where a terminal **#** is used, Pexip Infinity will automatically use the appropriate audio file, asking participants to "please enter the conference PIN number followed by the pound key". Preconfigured themes allow you to change the use of "pound key" to "hash key".

For Infinity Connect clients, the terminal **#** is optional when entering a PIN.

Examples

- If you want to use a terminal # on a VMR with a Host PIN of 1234 and a Guest PIN of 567890:
  - in the **Host PIN** field enter **1234#**
  - in the **Guest PIN** field enter **567890#**.

  Now when anyone calls the VMR they will hear the message "please enter the conference PIN number followed by the pound key".
- If you do not want to use a terminal # on a VMR with a Host PIN of 1234 and no Guest PIN:
  - in the **Host PIN** field enter **1234**
  - leave the **Guest PIN** field blank.

  Now when anyone calls the VMR they will hear the message "please enter the conference PIN number".

How to combine Host PINs and Guest PINs for differing levels of security

The table below shows how to use **Host PIN**s and **Guest PIN**s to achieve various levels of security for your Virtual Meeting Rooms and Virtual Auditoriums:

Access	Roles	Host PIN	Allow Guests	Guest PIN
Anyone - no PIN required	All participants have the same Host privileges.	Leave blank	Select *No*	Leave blank
All participants must enter the same PIN	All participants have the same Host privileges.	Enter the PIN	Select *No*	Leave blank
Hosts must enter a PIN but Guest do not	Participants who enter the PIN have Host privileges. The conference will not begin until they have joined, and will finish a minute or so after the last Host leaves. All other participants have Guest privileges and do not need to enter a PIN. They will see a holding screen until the first Host joins.	Enter the PIN	Select *Yes*	Leave blank
Hosts and Guests must enter different PINs	Participants who enter the **Host PIN** have Host privileges. The conference will not begin until they have joined, and will finish a minute or so after the last Host leaves. Participants who enter the **Guest PIN** have Guest privileges and will see a holding screen until the first Host joins.	Enter the PIN (must be different from the **Guest PIN**)	Select *Yes*	Enter the PIN* (must be different from the **Host PIN**)

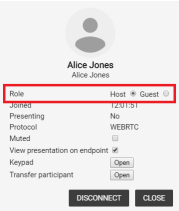
* If you configure a **Guest PIN**, you **must** also configure a **Host PIN**.

Changing a participant's role from Guest to Host (and vice versa)

Infinity Connect users who have a role of Host can change the role of other participants in the conference. This is useful, for example, if a Host needs to leave the conference; they can promote one of the Guests to a role of Host so that the conference will continue after they disconnect.

You can also use this feature to allow a participant who is waiting at the PIN entry screen to join the conference as Host or Guest without entering a PIN. Such users will appear in the roster with a role of **Unknown**. (Note that this does not apply to participants using Infinity Connect clients, as they use a different PIN entry process.)

To change a participant's role, select the participant and then select the role you wish to assign them:



You cannot change your own role to Guest.

Including the PIN in the dial string to bypass the PIN entry screen

SIP and H.323 endpoints and Lync / Skype for Business clients that dial into PIN-protected conferences can bypass the PIN entry screen by including the PIN in the dialed alias.

This makes it more convenient, for example, when bookmarking PIN-protected conferences.

They can do this by including the relevant PIN (either the Host PIN or the Guest PIN as appropriate) in their dial string when dialing the Virtual Meeting Room or Virtual Auditorium. The dial string should be in the format: **<vmr_alias>**<PIN>@<domain>**.

For example, if the alias of the Virtual Meeting Room is **meet.alice@example.com** and the PIN is **1234**, then the endpoint can dial **meet.alice**1234@** to go straight into the VMR.

Note that H.323 devices can also use the dial format **<vmr_alias>#<PIN>@<domain>**.

If there is a **#** configured at the end of the PIN e.g. **1234#**, then that **#** is not required in the dial string.

Playing notification tones when participants join or leave a conference

Pexip Infinity can automatically play a sound when a participant joins or leaves a conference. These entry and exit tones are disabled by default, but you can enable them on selected services using themes.

Enabling notification tones

Entry and exit tones are applied using themes. To enable entry or exit tones for a particular Virtual Meeting Room or Virtual Auditorium, you must ensure that the theme used for that service includes the following files:

- **conf-participant_entry_tone_48kHz_mono.wav**, containing the sound to play when a participant joins the conference.
- **conf-participant_exit_tone_48kHz_mono.wav**, containing the sound to play when a participant leaves the conference.

By default, entry and exit tones are not played, as the Base theme contains "empty" entry and exit tone files. However, Pexip Infinity also ships with a selection of preconfigured themes, some of which contain entry and exit tones in the **.wav** files listed above.

This means that to enable notification tones you can do any of the following:

- select one of the preconfigured themes containing tones
- use the **.wav** files from those preconfigured themes in your own themes
- produce your own **.wav** files and include those files in your own themes.

Disabling notification tones

The Base theme contains empty entry and exit tone files, and is shipped as the default theme. This means that by default entry and exit tones will not be played. You can download and copy the empty entry and exit tone files from the Base theme and use them in any custom themes you create for which you don't want to include entry and exit tones. Some of the preconfigured themes also contain empty entry and exit tone files.

If you apply a theme that does not include these files (rather than including empty files), Pexip Infinity will instead use the entry and exit tone files from the theme that has been selected as the default theme.

Therefore, if you want to ensure that entry or exit tones are not played, you must still include the entry tone and exit tone files in the theme, but leave them empty.

Selecting the layout seen by participants

A layout is the view that each participant has of all the other participants in the conference. Layouts are chosen in advance by the administrator, and some of the options available will depend on whether the conference is being held in a Virtual Meeting Room or Virtual Auditorium.

You cannot convert an existing Virtual Meeting Room into a Virtual Auditorium. Instead, you must create a new Virtual Auditorium and reassign the existing aliases to it. For more information, see Changing from a Virtual Meeting Room to a Virtual Auditorium and vice versa.

Features common to all layouts

Regardless of whether the conference is being held in a Virtual Meeting Room or Virtual Auditorium,

- Audio-only participants are represented by an avatar on the left side of the video window. The avatar will be either the participant's image (if available via an policy profiles) or a telephone icon:
- If there are more than 4 audio-only participants, they are represented by a single icon on the left side of the video window. This icon indicates the total number of audio-only participants:
- When an audio-only participant speaks, their avatar will slide out to show their name.

- Participants who are receiving video but not sending video (for example, if they have disabled their camera) are represented by an icon of a camera with a line through it .
- Video participants who are on hold, or who are experiencing connectivity issues, are represented by a frozen grayscale image.
- If there are more video participants than there are live thumbnail views available, the last thumbnail is replaced by an icon indicating the number of additional participants not currently visible.
- When a participant is the only device that is sending video, that participant sees a holding screen until other video participants join the conference. The holding screen indicates either that the participant is the only participant in the conference, or that all of the other participants are audio-only (and they are shown via their audio-only indicators as usual). The holding images are fully customizable via the theme associated with the service (Virtual Meeting Room, Call Routing Rule etc).
- A full list of participants is available using Infinity Connect clients.
- If the conference includes a streaming participant (e.g. it is being streamed to YouTube), a streaming icon is displayed to the right of the video window.
- When sending presentation content to a single-stream endpoint such as Skype or broadcasting via an RTMP stream (when not dialed out with dual streaming), Pexip Infinity sends the video stream of the active speaker in a small window in the upper right corner of the presentation.

Showing the names of speakers

Each Virtual Meeting Room and Virtual Auditorium has the option to show the name of the current speaker along the bottom of their video image. To enable this option, go to **Service Configuration > Virtual Meeting Rooms** or **Service Configuration > Virtual Auditoriums**, select the service, and next to the **Show names of speakers** field select ***Yes***.

When this option has been enabled, the names of the speakers will be shown as follows:

- Automatically dialed participants (regardless of the type of endpoint that was dialed): either the ADP's configured **Participant display name** or, if this field is blank, the **Participant alias**, will be shown.
- Manually dialed participants (regardless of the type of endpoint that was dialed): either the participant's configured **Participant display name** or, if this field is blank, the **Participant alias**, will be shown.
- Infinity connect desktop client and Web App participants: the text that the user entered into the client's **Your name** field will be shown.
- Infinity Connect Mobile client for Android: the text that the user entered into the client's **Profile** field will be shown
- Infinity Connect Mobile client for iOS: the text that the user configured in the client's **Connection Settings > Display Name** field will be shown. (If they haven't configured this field, the device's name will be shown instead.)
- SIP endpoints: the endpoint's display name will be shown.
- H.323 endpoints: the endpoint's alias will be shown.
- Lync / Skype for Business clients: the username portion of the user's sign-in address will be used, for example if Alice signs in as **alice@example.com** the text **alice** will be shown.

The default font for this and all other overlaid system-generated text is Roboto, or if that is not available for the selected language, Noto Sans.

Virtual Meeting Room layouts

All video-enabled participants in a Pexip Infinity Virtual Meeting Room see the same layout, regardless of whether they are Hosts or Guests. Administrators can select between the following options (go to **Service Configuration > Virtual Meeting Rooms** and select the Virtual Meeting Room you wish to change):

Option	Description
Full-screen main speaker only (1 + 0 layout)	Participants see a single speaker, with no thumbnails. • The current speaker is shown full-screen. (The current speaker sees the previous speaker.) • The number of other participants (Hosts plus Guests) is indicated in a small icon at the bottom right of the screen.

Option	Description

Large main speaker and up to 7 other participants (1 + 7 layout)	Participants see each other in the standard 1+7 layout.
	• The main video shows the current speaker (or previous speaker, for those currently speaking).
	• Up to 7 other participants are shown in a single row of live thumbnails at the bottom of the screen.
	  ○ Participants are shown in the thumbnails in order of who spoke most recently, from left to right, regardless of whether they are a Host or Guest.
	  ○ If there are more participants than there are thumbnails available, the number of additional participants is indicated in the far right thumbnail.

Small main speaker and up to 21 other participants (1 + 21 layout)	This is the same as the ***1 + 7 layout*** option, except the main speaker is shown slightly smaller in order to accommodate up to 21 other participants in 3 rows of live thumbnails at the bottom of the screen.
2 small main speakers and up to 21 other participants (2 + 21 layout)	This is the same as the ***1 + 21 layout*** option, except the top row contains both the main speaker and most recent speaker side by side.

Virtual Auditorium layouts

With Pexip Infinity Virtual Auditoriums, Guest participants only ever see the Host participants; however, Hosts will see Hosts and Guests. Administrators choose from separate options for the layout shown to Guests and the layout shown to Hosts. Additionally, administrators can configure the Virtual Auditorium so that when a presentation is being shown, the presenter is kept in the main view.

To select which layouts are shown to Hosts and Guests, go to **Service Configuration > Virtual Auditoriums** and select the Virtual Auditorium you wish to change. The options available are described below.

Host view

Option	Description
Full-screen main speaker only (1 + 0 layout)	Hosts see a single speaker, with no thumbnails. • The current Host speaker is shown full-screen. (The currently speaking Host sees the previous Host speaker, or if there are no other Hosts, they see the most recent Guest speaker.) • The number of other participants (Hosts plus Guests) is indicated in a small icon at the bottom right of the screen.

Large main speaker and up to 7 other participants (1 + 7 layout)	Hosts see other Host and Guest participants in the standard 1+7 layout.
	• If there is only one Host, they see the most recent Guest speaker in the main video.
	• If there is more than one Host, they see the currently speaking Host in the main video. (The currently speaking Host sees the previous Host speaker.)
	• Up to 7 other participants are shown in a single row of live thumbnails at the bottom of the screen.
	  ○ Host participants are shown first in the thumbnails, in order of who spoke most recently, from left to right.
	  ○ Any remaining thumbnails are used to show Guests (again, in order of who spoke most recently).
	  ○ If there are more participants than there are thumbnails available, the number of additional participants is indicated in the far right thumbnail.

Small main speaker and up to 21 other participants (1 + 21 layout)	This is the same as the ***1 + 7 layout*** option, except the main speaker is shown slightly smaller in order to accommodate up to 21 other participants in 3 rows of live thumbnails at the bottom of the screen.
2 small main speakers and up to 21 other participants (2 + 21 layout)	This is the same as the ***1 + 21 layout*** option, except the top row contains both the main speaker and most recent speaker side by side. This option is useful in combination with the Lock presenter as main speaker option, so that other participants can see the presenter and the person they are speaking to at the same time.

Guest view

Guests in a Virtual Auditorium are able to:

- hear and see the Host participant(s)
- hear but not see any of the other Guests, even if a Guest is speaking.

However, if all Hosts have left the conference, Guests will be able to see and hear other Guests in the selected layout until the conference is automatically disconnected (after about a minute), or until a Host rejoins.

Option	Description
Full-screen Host speaker only (1 + 0 layout)	Guests see a single Host speaker, with no thumbnails. • The Host who spoke most recently is shown full-screen. • If there are two or more Hosts, the number of other Host participants is indicated in a small icon at the bottom right of the screen.
Large Host speaker and up to 7 other Hosts (1 + 7 layout)	Guests see Host participants in the standard 1+7 layout. • If there is only one Host participant, the Host is shown full-screen. • If there is more than one Host participant, the standard 1+7 layout is used, with the main video showing the currently speaking Host and all other Host participants shown in the thumbnails. • Hosts are shown in the thumbnails in order of who spoke most recently, from left to right. • If there are more Host participants than there are thumbnails available, the number of additional Hosts is indicated in the bottom right thumbnail.
Small Host speaker and up to 21 other Hosts (1 + 21 layout)	This is the same as the ***1 + 7 layout*** option, except the main Host speaker is shown slightly smaller in order to accommodate up to 21 other Hosts in 3 rows of live thumbnails at the bottom of the screen.
2 small Host speakers and up to 21 other Hosts (2 + 21 layout)	This is the same as the ***1 + 21 layout*** option, except the top row contains both the main Host speaker and most recent Host speaker side by side. This option is useful in combination with the Lock presenter as main speaker option, so that other participants can see the presenter and the person they are speaking to at the same time.

Lock presenter as main speaker

You can configure a Virtual Auditorium so that when a presentation is being shown, the main speaker position always shows the presenter instead of the current speaker.

Option	Description
Yes	When a presentation is being shown, the selected layout rules are overridden so that: • The main speaker position always shows the video image from the endpoint that is showing the presentation, even if others are speaking. • The image that would have been shown in the main view is instead shown in the first available thumbnail (for 1 + 7 and 1 + 21 layouts) or in the top right view (for 2 + 21 layout).
No	When a presentation is being shown, the main speaker position will be voice-switched as usual.

1 + 0 layout

1 + 0 layout: full-screen main video plus an icon indicating the number of additional participants

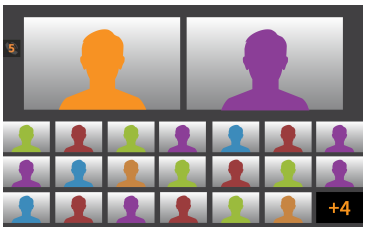
1 + 7 layout

standard 1+7 layout: main video plus a single row of thumbnails, an indicator showing the number of additional video participants, and two icons representing the audio-only participants (one with a personal avatar).

1+21 layout

1+21 layout: main video plus three rows of thumbnails, an indicator showing the number of additional video participants, and an indicator showing the total number of audio-only participants.

2 + 21 layout



2 + 21 layout: two main speakers plus three rows of thumbnails, an indicator showing the number of additional video participants, and an indicator showing the total number of audio-only participants.

Automatically dialing out to a participant from a conference

You can configure a Virtual Meeting Room or Virtual Auditorium to automatically dial out to one or more participants whenever a conference using that service starts.

You can also **manually** dial out to participants from a Virtual Meeting Room or Virtual Auditorium, on an ad-hoc basis. For more information, see Manually dialing out to a participant from a conference.

When will the participant be called?

When an Automatically Dialed Participant (ADP) has been added to a Virtual Meeting Room or Virtual Auditorium's configuration, a call is placed from the conference to the ADP's endpoint as follows:

Meeting type	Participant automatically dialed...
No PIN	when the first participant joins the conference
Guests not allowed; participants must enter a PIN	when the first participant has entered a valid PIN
Guests allowed; Hosts must enter a PIN but Guests do not	when the first participant joins the conference
Guests allowed; Hosts and Guests must enter a PIN	when the first participant has entered a valid PIN

The role of the first participant does not matter - both Hosts and Guests can trigger a call to an ADP.

A call is not placed to an ADP if that participant has already joined the conference.

Host and Guest PINs

When the call is answered, the Automatically Dialed Participant will join the conference as either a Host or Guest, depending on which role you selected when configuring them as an ADP. They will not be required to enter a PIN even if one is required for that role.

Choosing the calling location

Usually, the call to the ADP is placed from the same Conferencing Node that is being used by the participant who initiated the conference.

However, you can optionally configure the call to be placed from a Conferencing Node in a specific location. The behavior varies according to whether it is a manually routed ADP (where **Route this call** is set to ***Manually***) or the ADP is called according to Call Routing Rules (where **Route this call** is set to ***Automatically***).

- **Manually routed ADPs**: the nominated location is the location of the node that will dial the ADP.
- **Automatically routed ADPs**: the nominated location is the notional source location used when considering if a Call Routing Rule applies or not — but the rules themselves will determine the location of the node that dials the ADP. In this case, the configured **Outgoing location** for the ADP is deemed to be the location handling the call (when matched against the **Calls being handled in location** setting for the Call Routing Rule) and the rule's **Outgoing location** setting is used to determine the actual location of the node that dials the ADP.

  This behavior is useful if, for example, you have configured lowest-cost-routing rules that only allow calls to be placed from "internal" Conferencing Nodes but you have a conference that is triggered by a call to an "external" Conferencing Node. In this case you still want the call to the ADP to be initiated, but you don't want, in general, to allow calls to arbitrary ISDN destinations to be placed from Conferencing Nodes in the external location.

Incoming call alias

When the endpoint is called, the Automatically Dialed Participant will see the call as coming from one of the Virtual Meeting Room or Virtual Auditorium's aliases. This means that if the participant misses the call, they can easily return it by dialing the alias that appears on their endpoint.

When deciding which alias to use to identify the call, Pexip Infinity will select the first in the list that is valid for the selected call protocol. So if your Virtual Meeting Room has two aliases, the first **meet.sales** and the second **meet.sales@example.com**, when you place a call to an ADP over SIP it will show as coming from **meet.sales** because that is the first valid SIP address in the list. However, endpoints may not always be able to return a call to this alias because it is missing the domain. Therefore, if you want ADPs to be able to return calls to Virtual Meeting Rooms and Virtual Auditoriums, we recommend that you list the most routable aliases first when assigning them to these services. Note that you can also use external or local policy (see policy profiles) to specify the source alias for calls to ADPs.

Keeping a conference alive

Automatically Dialed Participants with a role of Host may or may not prevent a conference from being automatically terminated, depending on their **Keep conference alive** setting and whether any other ADPs remain in the conference. For more information, see Automatically ending a conference.

Configuration

If you have a large number of ADPs to configure, you can import them using a CSV file. For more information, see Bulk import/export of Automatically dialed participants (ADPs).

To configure a Virtual Meeting Room or Virtual Auditorium to automatically dial out to a participant when a conference starts:

1. Go to **Service Configuration > Automatically Dialed Participants**.
2. Select **Add Automatically Dialed Participant**.
   You will be taken to the **Add Automatically Dialed Participant** page.
3. Complete the following fields:

Field	Description
Participant alias	The alias of the participant to dial when a conference starts. If you select a **Protocol** of ***SIP***, this must be a valid SIP alias.
Creation time	This read-only field shows the date and time at which this record was first configured.
Participant display name	An optional user-facing display name for this participant, which may be used in participant lists and as the overlaid speaker name (if enabled). If this name is not specified then the **Participant alias** is used as the display name instead.
Description	An optional description of the Automatically Dialed Participant.
Route this call	Select how to route the call: ○ ***Manually***: uses the requested **Protocol** and the defaults for the specified **System location**. ○ ***Automatically***: routes the call according to the configured Call Routing Rules— this means that the dialed alias must match an outgoing Call Routing Rule for the call to be placed (using the protocols and call control systems etc. as configured for that rule). Default: ***Manually***.
Protocol	The signaling protocol to use when dialing the participant. Select either ***SIP***, ***H.323***, or if the endpoint is a Lync / Skype for Business client, select ***Lync / Skype for Business (MS-SIP)***. The RTMP protocol is typically used when adding a streaming participant. Note that if the call is to a registered device, Pexip Infinity will instead use the protocol that the device used to make the registration. Default: ***SIP***. This field only applies when **Route this call** is set to ***Manually***.
DTMF sequence	An optional DTMF sequence to be transmitted after the call to the Automatically Dialed Participant starts. A DTMF sequence can include: ○ the digits 0-9 ○ * (asterisk) ○ # (hash) ○ , (comma). The DTMF tones will be sent 3 seconds after the call connects, one at a time, every 0.5 seconds. A comma is a special digit that represents a 2 second pause (multiple commas can be used if a longer pause is needed). For example, if you need your Automatically Dialed Participant to dial an audio bridge and then enter conference number 666 followed by #, pause for six seconds and then supply conference PIN 1234 followed by #, you would configure **666#,,,1234#** as your DTMF sequence.

Field	Description
Role	The level of privileges the participant will have in the conference. For more information, see About PINs, Hosts and Guests. Default: ***Guest***.
Conference	Select the names of the Virtual Meeting Rooms and Virtual Auditoriums from which this participant will be dialed automatically whenever a conference using that service starts.
Outgoing location	For ***Manually*** routed ADPs, this is the location of the Conferencing Node from which the call to the ADP will be initiated. For ***Automatically*** routed ADPs, this is the notional source location used when considering if a routing rule applies or not - however the routing rule itself determines the location of the node that dials the ADP. For more information, see Choosing the calling location. To allow Pexip Infinity to automatically select the Conferencing Node to use to place the outgoing call, select ***Automatic***. As with all calls, signaling and media may be handled by different Conferencing Nodes in that location.
Streaming	Identifies the dialed participant as a streaming or recording device. See Streaming a conference over YouTube for more information.
Keep conference alive	Determines whether the conference will continue when all other non-ADP participants have disconnected. ***Yes***: the conference will continue to run until this participant has disconnected (applies to Hosts only). ***If multiple***: the conference will continue to run as long as there are two or more ***If multiple*** participants and at least one of them is a Host. ***No***: the conference will be terminated automatically if this is the only remaining participant. Default: ***If multiple***. For streaming participants, we recommend that this option is set to ***No***. For more information, see Automatically ending a conference.
Call capability	Allows you to limit the media content of the call. The participant being called will not be able to escalate beyond the selected capability. For more information, see Controlling media capability. Default: ***Main video + presentation***. This field only applies when **Route this call** is set to ***Manually***. This field does not apply to RTMP calls.
Dual stream (presentation) URL	When adding a dual streaming RTMP participant, this specifies the RTMP URL for the second (presentation) stream. Leave this field blank when adding a single streaming participant.

4. Select **Save**.

Manually dialing out to a participant from a conference

The Pexip Infinity platform allows you to manually dial out to participants from a conference, on an ad-hoc basis. When you dial out to a participant in this way, a call is placed to their endpoint from the Virtual Meeting Room or Virtual Auditorium. If and when they answer the call, they will join the conference as either a Host or Guest, depending on the option that you selected. Participants joining the conference in this way will not go through the IVR screen and will not have to enter a PIN.

The participant could also take the form of a dedicated multimedia stream to enterprise CDN (Content Delivery Network) streaming and recording services such as Wowza, Adobe, VBrick, Abiliteam, Qumu and Azure Media Services, and to public streaming services such as YouTube. Any Pexip conference can be streamed as a live event to an unlimited number of viewers, and can automatically be recorded and stored for later consumption. For more information, see Streaming a conference over YouTube.

Call Routing Rules may be applied when dialing out from a conference to a new participant:

- Call Routing Rules are applied if **automatic** routing is selected when placing the call — this means that the dialed alias must match an outgoing Call Routing Rule for the call to be placed (using the protocols and call control systems etc. as configured for

that rule).

- For **manual** routing, Pexip Infinity always attempts to route the call according to the requested protocol and media capabilities.

The ability to manually dial out to a participant is enabled by default. You can disable this setting by going to the **Global Settings** page (**Platform Configuration > Global Settings**) and clearing the **Enable outbound calls** checkbox (note that this will also disable calls made via the Pexip Distributed Gateway).

Both Administrators and conference hosts can manually add a participant to a conference. Administrators can do so Using the Administrator interface, and hosts can do so Using Infinity Connect.

You can also configure a Virtual Meeting Room or Virtual Auditorium so that one or more participants are dialed out to automatically whenever a conference starts. For more information, see Automatically dialing out to a participant from a conference.

If your environment includes a PSTN gateway or uses an ITSP (Internet telephony service provider), consider the potential for toll fraud if you have Call Routing Rules that can route calls to the PSTN gateway or ITSP, or if you allow conference participants to dial out to other participants via the PSTN gateway or ITSP. See PSTN gateways and toll fraud for more information.

Using the Administrator interface

You can use the Pexip Infinity Administrator interface to dial out to a participant from a Virtual Meeting Room or Virtual Auditorium. If and when the call is answered, that endpoint will join the conference.

To dial out to a participant using the Administrator interface:

1. Select the Virtual Meeting Room or Virtual Auditorium to dial the participant from. You can do this in any of the following ways:
   - Go to **Service Configuration > Virtual Meeting Rooms** and select the name of the Virtual Meeting Room.
   - Go to **Service Configuration > Virtual Auditoriums** and select the name of the Virtual Auditorium.
   - If the conference that you wish to dial out from is already in progress, go to **Status > Conferences** and select the name of the Virtual Meeting Room or Virtual Auditorium being used.
2. At the bottom left of the screen, select **Dial out to participant**.
3. Complete the following fields:

Field	Description
System location	Select the system location from which the call will be placed. If there is more than one Conferencing Node in that location, Pexip Infinity will choose the most appropriate. The system location determines which H.323 gatekeeper or SIP proxy to use to route the call.
Service alias	This lists all the aliases that have been configured for the selected Virtual Meeting Room or Virtual Auditorium. The participant will see the incoming call as coming from the selected alias.
Participant alias	The alias of the endpoint/participant that you want to dial.
Route this call	Select how to route the call: ◦ ***Manually***: uses the requested **Protocol** and the defaults for the specified **System location**. ◦ ***Automatically***: routes the call according to the configured Call Routing Rules— this means that the dialed alias must match an outgoing Call Routing Rule for the call to be placed (using the protocols and call control systems etc. as configured for that rule). Default: ***Manually***.
Participant display name	An optional user-facing display name for this participant, which may be used in participant lists and as the overlaid speaker name (if enabled). If this name is not specified then the **Participant alias** is used as the display name instead.

Field	Description
Protocol	The signaling protocol to use when dialing the participant. Select either ***SIP***, ***H.323***, or if the endpoint is a Lync / Skype for Business client, select ***Lync / Skype for Business (MS-SIP)***. The RTMP protocol is typically used when adding a streaming participant. Note that if the call is to a registered device, Pexip Infinity will instead use the protocol that the device used to make the registration. This field only applies when **Route this call** is set to ***Manually***.
Call capability	Allows you to limit the media content of the call. The participant being called will not be able to escalate beyond the selected capability. For more information, see Controlling media capability. Default: ***Main video + presentation***. This field only applies when **Route this call** is set to ***Manually***.
Role	Select whether you want the participant to join the conference as a Host or Guest.
DTMF sequence	An optional DTMF sequence to be transmitted after the call to the dialed participant starts. A DTMF sequence can include: ○ the digits 0-9 ○ * (asterisk) ○ # (hash) ○ , (comma). The DTMF tones will be sent 3 seconds after the call connects, one at a time, every 0.5 seconds. A comma is a special digit that represents a 2 second pause (multiple commas can be used if a longer pause is needed). For example, if you are dialing an audio bridge and want to enter conference number 666 followed by #, pause for six seconds and then supply conference PIN 1234 followed by #, you would configure **666#,,,1234#** as your DTMF sequence.
Streaming	Identifies the dialed participant as a streaming or recording device. See Streaming a conference over YouTube for more information.
Dual stream (presentation) URL	When adding a dual streaming RTMP participant, this specifies the RTMP URL for the second (presentation) stream. Leave this field blank when adding a single streaming participant.
Keep conference alive	Determines whether the conference will continue when all other participants have disconnected. ***Yes***: the conference will continue to run until this participant has disconnected (applies to Hosts only). ***If multiple***: the conference will continue to run as long as there are two or more ***If multiple*** participants and at least one of them is a Host. ***No***: the conference will be terminated automatically if this is the only remaining participant. Default: ***Yes***. For streaming participants, we recommend that this option is set to ***No***. For more information, see Automatically ending a conference.

4. Select **Dial out to participant**.

A message **Initiated dial out to participant** will appear at the top of the screen.

To confirm whether the participant has joined the conference:

1. Go to **Status > Conferences**.
2. Select the name of the Virtual Meeting Room or Virtual Auditorium.
3. Select the **Participants** tab.

The participant will appear in the list.

Using Infinity Connect

If you have Host privileges, you can use the Infinity Connect desktop client, the Infinity Connect Web App, or the Infinity Connect Mobile client for Android to dial out to participants from the Virtual Meeting Room or Virtual Auditorium you are in. When you use

this feature, a call is placed to the participant and if and when they answer the call, they will join the conference directly (they will not go through the IVR screen and will not have to enter a PIN).

To dial out to a participant using Infinity Connect, from within a Virtual Meeting Room or Virtual Auditorium:

1. Select the menu icon and then select **Add a new participant**.
2. At the prompt, enter the address of the person you want to dial.
3. If you want to use a protocol other than ***SIP*** (the default) select either ***Automatic***, ***H.323***, ***Lync/Skype*** or ***RTMP***.

   Automatic means that the protocol will be selected according to how your administrator or service provider has configured the system.

   RTMP is typically used when connecting to a streaming or recording service.

   To support the use of ***Automatic*** you must configure some appropriate Call Routing Rules that apply to **Outgoing calls from a conference**. If required, you can then also change the default protocol presented to users of the Infinity Connect Web App and desktop clients by configuring the **defaultDialOutProtocol** setting in the **settings.js** file (see Advanced Infinity Connect Web App customization and Customizing the Infinity Connect desktop client for more information).
4. Select whether you want the participant to have **Host** or **Guest** privileges.
5. Select **OK**.

A call will be placed from the Virtual Meeting Room or Virtual Auditorium to the participant and they will appear in the participant list with the "calling" icon next to their avatar while their endpoint is ringing. If and when the participant answers the call they will join the conference; if they do not answer, or do not accept the call, they will disappear from the participant list.

Using the Infinity Connect Mobile client for iOS

In order to use this feature:

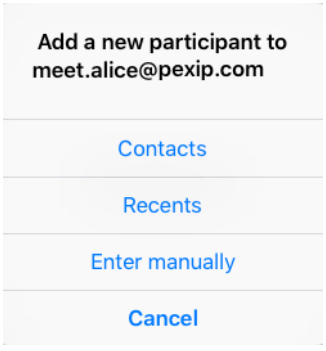
- You must have Host privileges.
- In some cases, your administrator must have enabled this functionality. See Enabling and disabling SIP, H.323, WebRTC and RTMP for more information.
- You must have agreed to allow the Infinity Connect Mobile client to access your contacts.

You can use the Infinity Connect Mobile client to dial out to participants from the Virtual Meeting Room or Virtual Auditorium you are in. When you use this feature, a call is placed to the participant and if and when they answer the call, they will join the conference as a Guest participant (or as a normal participant, if the conference has no PIN). Participants joining the conference in this way will not go through the IVR screen and will not have to enter a PIN.

To dial out to a participant using the Infinity Connect Mobile client for iOS when in a call:

1. From the left of the screen, select and then **Add a new participant**.

   The following screen will appear:



2. Select from the available options:
   - to call someone already in your Contacts, select **Contacts**
   - to call someone you have recently called from the Pexip App, select **Recents**

- to call anyone else, select **Enter Manually**. In the **Address** field enter the number or URI of the participant, and then select the **Protocol**. (Note: ***Automatic*** means that the protocol will be selected according to how your administrator or service provider has configured the system.)

A call will be placed from the Virtual Meeting Room or Virtual Auditorium to the participant. If and when the participant answers the call they will join the conference and appear on the participant list.

Transferring a participant to another conference

You can transfer a participant from one conference to another conference (or to another device or system via the Pexip Distributed Gateway) in three ways: Using the Administrator interface, Using Infinity Connect clients, or Using the APIs.

After the transfer has been successfully initiated, the transferred participant will be removed from the original conference. Should the transfer then fail (for example, if it is made to a device that rejects the call), the transferred participant will be disconnected i.e. it will not rejoin the original conference.

Note that when a Lync / Skype for Business participant is transferred, they will appear as audio-only and will need to escalate to video.

Using the Administrator interface

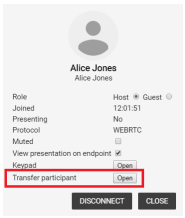
To use the Pexip Infinity Administrator interface to transfer a single participant from one conference to another conference:

1. Select the participant. You can do this in two ways:
   - Go to **Status > Participants** and select the participant to transfer.
   - Go to **Status > Conferences** and select the Virtual Meeting Room or Virtual Auditorium that the participant is currently in. From the **Participants** tab, select the participant to transfer.
2. At the bottom right of the screen, select **Transfer**.
   The **Transfer Participant** dialog opens.
3. Enter the **Target conference alias**.
   This can be any alias associated with a Virtual Meeting Room, Virtual Auditorium or Call Routing Rule.
4. Select the **Role** the participant will have in the new conference.
5. Select **Transfer**.
   If the transfer is successful, the participant list will be updated to show the participant in the new conference (it may also temporarily show the participant still connected to the previous conference).

Using Infinity Connect clients

To use the Infinity Connect desktop client or Infinity Connect Web App to transfer a participant (including yourself) from one conference to another:

1. Join the conference as a Host.
2. From the participant list, select the participant's name. Next to **Transfer participant** select **Open**:



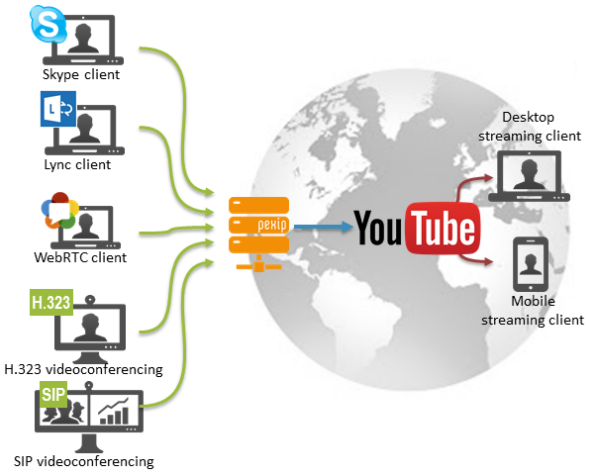
3. Enter the alias of the conference you wish to transfer the participant to, whether they should join as a ***Host*** or ***Guest***, and the PIN if applicable, then select **OK**.

Using the APIs

Third-party applications can use the management API and the client APIs to transfer participants to another conference.

For more information, see Management command API and Using the Pexip client APIs.

Streaming a conference over YouTube



Pexip Infinity can output a dedicated RTMP multimedia stream to enterprise CDN (Content Delivery Network) streaming and recording services such as Wowza, Adobe, VBrick, Abiliteam, Qumu and Azure Media Services, and to public streaming services such as YouTube. Any Pexip conference can be streamed as a live event to an unlimited number of viewers, and can automatically be recorded and stored for later consumption.

If your conference contains presentation content, you have the option of setting up dual streams so that you can output the main video and presentation content channels separately. This means that viewers can simultaneously access the main video stream of the participants and a separate presentation content stream.

The administrator or meeting Host has to decide whether to set up the conference with a single RTMP stream or with dual RTMP streams.

When a conference is single streamed:

- Viewers are automatically switched between the main video stream and the presentation stream whenever someone is presenting.
- When someone is presenting, the video stream of the active speaker is shown in a small window in the upper right corner of the presentation.
- The same single stream (with automatic switching between main video and presentation streams) can be made available later as a recording.

When a conference is dual streamed:

- Viewers have to manually switch between the main video stream and the presentation stream, according to their preference. (However, viewers could optionally view the streams in two separate windows — one window showing the main video stream and the other window showing the presentation stream.)
- The main video stream is not embedded as a small window within the presentation stream.
- If nobody is currently presenting, Pexip Infinity sends a placeholder image on the presentation stream.
- If the event is made available later as a recording, it can only be played back as two completely separate streams. There is no synchronization between the streams.

Note that your firewall needs to allow outbound traffic from the Conferencing Node to TCP port 1935 on the RTMP streaming server.

This guide explains how an administrator or meeting Host can stream a Pexip conference over YouTube.

YouTube prerequisites

Before the administrator or meeting Host can obtain an RTMP streaming URL from YouTube, they must ensure that they have a verified YouTube account, and that the account is enabled for live events.

1. From your YouTube account settings page, select **View additional features** (www.youtube.com/features).
2. If your **Account status** is not verified, select **Verify** and follow the YouTube instructions.
3. If **Live events** is not enabled, click **Enable**.

Setting up streaming

To stream a conference you must assign a live streaming URL via YouTube, and then add it as a participant in your Pexip conference. There are two ways in which you can obtain an RTMP streaming URL from YouTube:

- Use Pexip's own service at http://youtube.pexip.com. This method simplifies the generation process and automatically uses the appropriate settings. However, it is only suitable if you want to set up a **single stream**.

- Generate your URL directly from within your YouTube account at www.youtube.com/my_live_events. You must use this method if you want to set up **dual streams** within the same broadcast (i.e. a "second camera" in YouTube terms).

  When generating a URL directly from within YouTube, the **Privacy** setting is ***Public*** by default, so we recommend that you change this to ***Unlisted***.

Note that the live stream will have a 20-30 second delay. This is because YouTube buffers the stream so that it can tolerate brief connection losses and to ensure a good consistent experience. This is standard streaming behavior.

The appropriate procedures for adding single or dual-streamed participants to your conference are described below.

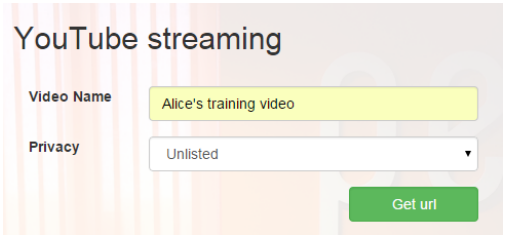
Setting up and adding a single streaming participant

This procedure explains how to use Pexip's own service to request a URL for a single YouTube RTMP stream on your behalf, and how to add that stream as a participant to your Pexip conference.

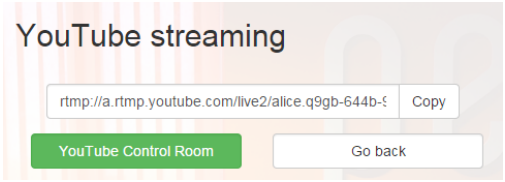
Obtaining the YouTube streaming URL

1. Go to http://youtube.pexip.com.
2. Enter a **Video Name** — this is the name that will appear in YouTube.
3. Select a **Privacy** level:
   - ***Unlisted***: viewers must know the streaming URL to see the stream.
   - ***Public***: anybody can find the stream on YouTube. This is not recommended unless you are streaming very public content.
   - ***Private***: restricts access to only people that you have explicitly allowed to view the stream.

   Default: ***Unlisted***.
4. Select **Get url**.



5. If you are not already signed in to a Google Account, you must either sign in or select an account.
6. The streaming URL will be generated and displayed.

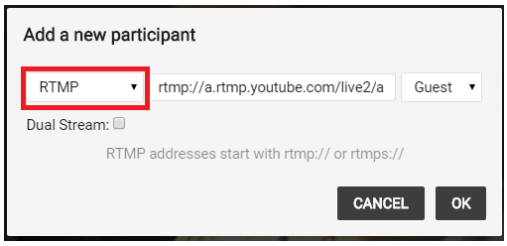


   If you receive a "The user is not enabled for live streaming" error message, this means that you either do not have a verified YouTube account, or that the account is not enabled for live events.
7. Select **Copy** to put the rtmp:// address onto your clipboard. Leave this browser window open.

Adding the participant URL and enabling streaming

Now that you have the YouTube streaming URL, you can add it as a participant in your Pexip conference and start streaming.

1. Add the YouTube streaming URL as a new RTMP participant in the Virtual Meeting Room you want to stream.
   a. Join the conference via Infinity Connect.
      (Note that administrators can also use the Pexip Infinity Administrator interface to add RTMP streaming participants.)
   b. Select **Add a new participant**.
   c. Select a protocol of ***RTMP***.
   d. Paste the streaming URL as the address to dial.



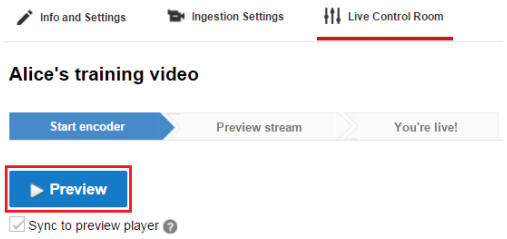
e. Select a role. We recommend selecting ***Guest*** (so that the streaming participant is not shown to other Guests in a Virtual Auditorium layout, and so that it does not keep a conference alive when all other Hosts have left).

f. Select **OK**.

Pexip Infinity will dial out to YouTube. The streaming participant will appear in the participant list with a streaming badge and a name that is typically in the format **a.rtmp.youtube.com**, and a streaming icon is displayed to the right of the layout.

Note that administrators can also add a streaming participant via the Administrator interface (go to **Status > Conference**, select the conference and then select **Dial out to participant**).

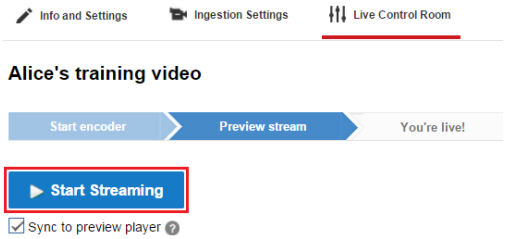
2. Wait for a few seconds, then from within your YouTube account, go to your **Live Control Room**. (You can select the green link below the rtmp:// address that you copied when using Pexip's URL generator.)
3. Select **Preview** and confirm.



After a few seconds you will be able to **Play** the Preview stream.

Note that this is your preview only — at this stage the stream is not being broadcast. The stream has a 20-30 second delay.

4. Click **Start streaming** and confirm, to start broadcasting.



5. You are now streaming to anyone who is allowed to access or find your streams (according to your **Privacy** settings). You can optionally **Play** the **Public View** of the stream.
6. You can select **View on Watch Page** (top right of your **Live Control Room** page) to see the normal YouTube view. This is how it appears to users who are watching the live stream and is the URL that you should share.

Setting up and adding a dual streaming participant

This procedure explains how to use YouTube to generate a YouTube streaming URL with dual streams, and how to add that stream as a participant to your Pexip conference.

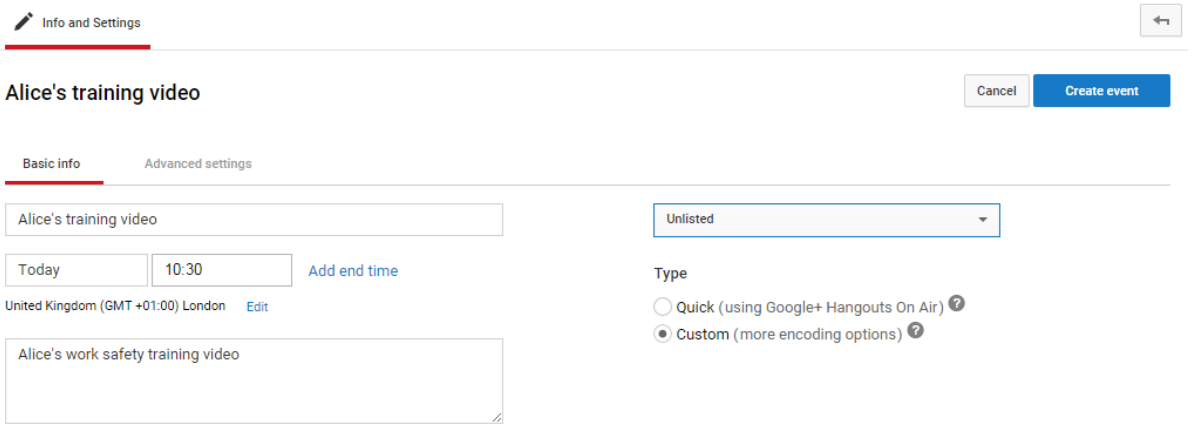
Note that the YouTube **Privacy** setting is ***Public*** by default, so we recommend that you change this to ***Unlisted***.

Obtaining the YouTube streaming URL

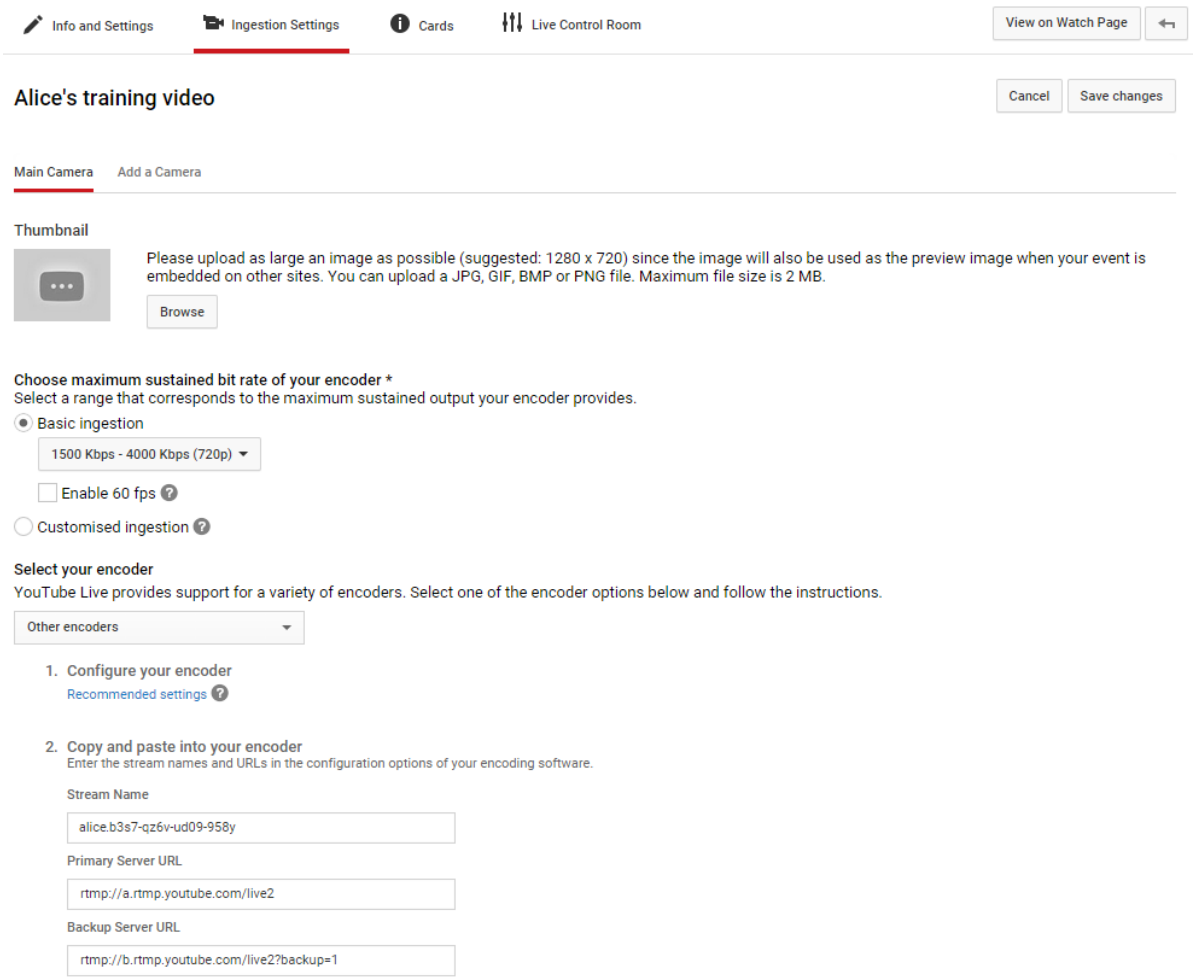
To generate your URL directly from within YouTube:

1. In your YouTube account go to **Video Manager > Live Events** (www.youtube.com/my_live_events).
2. Select **New live event** (at the top right of the page).
3. Enter the **Basic info**:

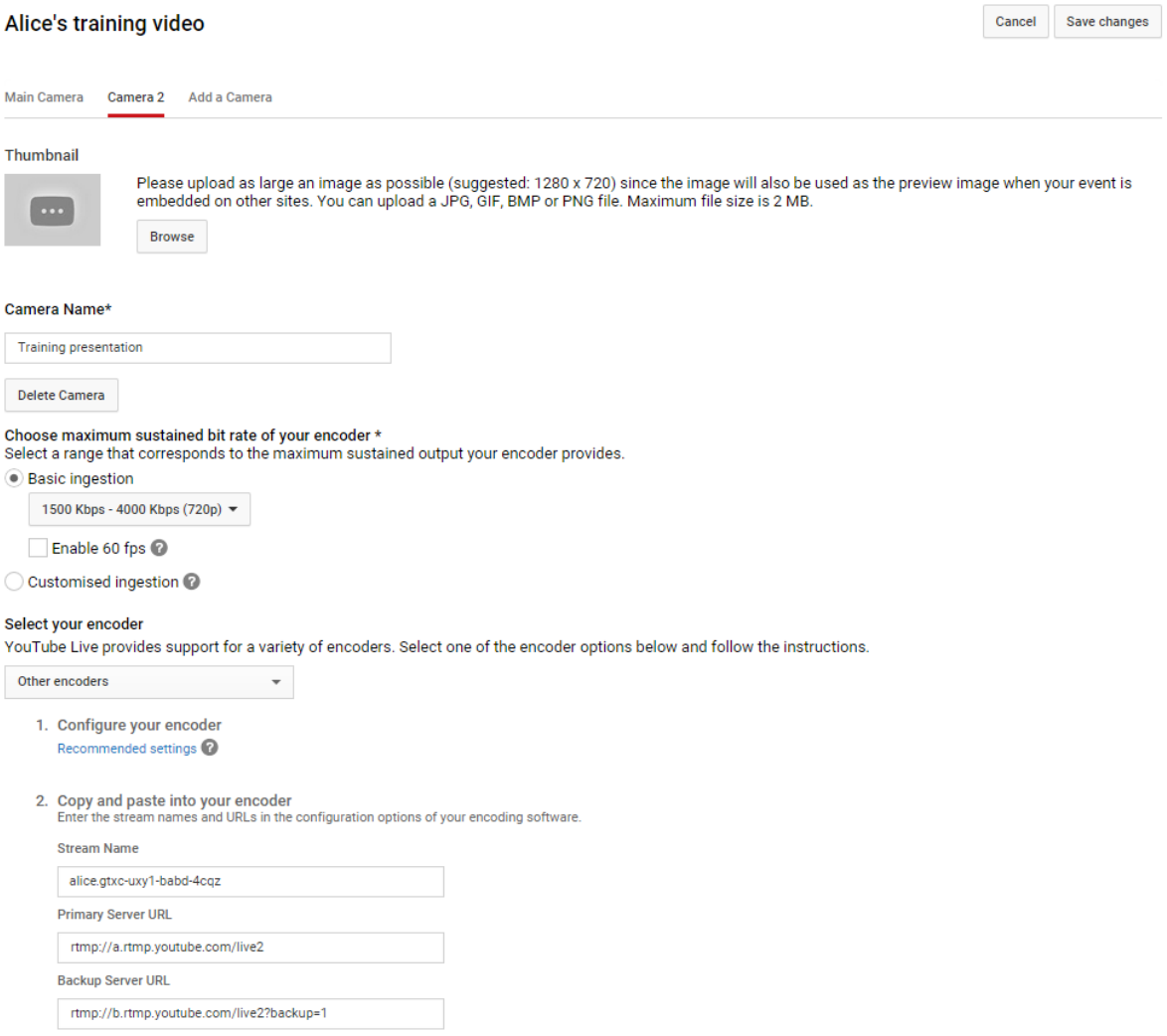
Title	The title of the video.
Start time	Set the start time to 30 minutes in the past (so that it is available instantly).
Privacy	Select a privacy level. We recommend ***Unlisted***, which means that viewers must know the streaming URL to see the stream.
Type	Select ***Custom***.



4. Select **Create event**. You are taken to the **Ingestion Settings** tab.
5. Configure the main video stream:
   a. Select ***Basic ingestion*** and select a bit rate of ***1500 Kbps - 4000 Kbps (720p)***.
   b. From the **Select your encoder** drop-down list, select ***Other encoders***.
   c. Optionally, you can upload a **Thumbnail** for this stream.



6. Configure the second stream:
   a. Select **Add a Camera** (next to the Main Camera tab).
   b. Enter the **Camera Name** e.g. "Training presentation".
   c. Select ***Basic ingestion*** and select a bit rate of ***1500 Kbps - 4000 Kbps (720p)***.
   d. From the **Select your encoder** drop-down list, select ***Other encoders***.
   e. Optionally, you can upload a **Thumbnail** for this stream.



7. Select **Save changes**.
8. Produce the RTMP URL for your primary video stream as **Primary Server URL/Stream Name** of the Main Camera:
   a. Go to the **Main Camera** tab.
   b. Take the **Primary Server URL**, for example **rtmp://a.rtmp.youtube.com/live2**.
   c. Append a **/** (slash).
   d. Then append the **Stream Name**, for example **alice.b3s7-qz6v-ud09-958y**.

   In this example, the RTMP URL is **rtmp://a.rtmp.youtube.com/live2/alice.b3s7-qz6v-ud09-958y**.

   You will use this RTMP URL as the first address of the new participant in your Pexip conference.
9. Produce the RTMP URL for your presentation stream as **Primary Server URL/Stream Name** of Camera 2:
   a. Go to the **Camera 2** tab.
   b. Take the **Primary Server URL**, for example **rtmp://a.rtmp.youtube.com/live2**.
   c. Append a **/** (slash).
   d. Then append the **Stream Name**, for example **alice.gtxc-uxy1-babd-4cqz**.

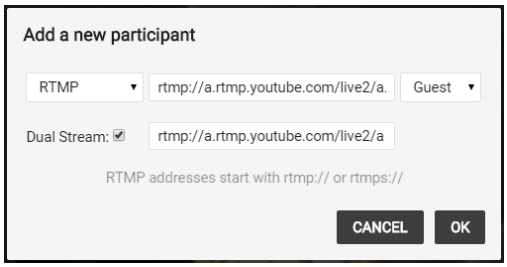
   In this example, the RTMP URL is **rtmp://a.rtmp.youtube.com/live2/alice.gtxc-uxy1-babd-4cqz**.

   You will use this RTMP URL as the second stream address of the new participant in your Pexip conference.

Adding the participant URL and enabling streaming

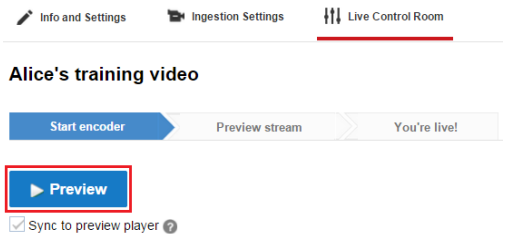
Now that you have the dual YouTube streaming URLs, you can add them as a participant in your Pexip conference and start streaming.

1. Using Infinity Connect, add the YouTube streaming URL as a new RTMP participant in the Virtual Meeting Room you want to stream.
   a. Join the conference via Infinity Connect.
   (Note that administrators can also use the Pexip Infinity Administrator interface to add RTMP streaming participants.)
   b. Select **Add a new participant**.
   c. Select a protocol of ***RTMP***.
   d. Paste the streaming URL of the primary video stream (from the **Main Camera** tab in the YouTube Ingestion settings) as the address to dial.
   e. Select **Dual Stream** and paste the second presentation streaming URL (from the **Camera 2** tab) into the address field that appears.

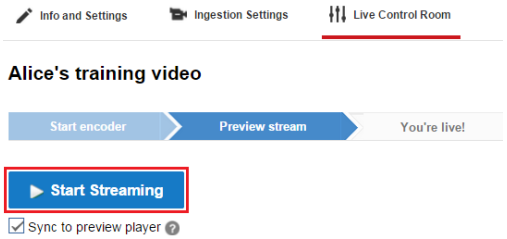


   f. Select a role. We recommend selecting ***Guest*** (so that the streaming participant is not shown to other Guests in a Virtual Auditorium layout, and so that it does not keep a conference alive when all other Hosts have left).
   g. Select **OK**.

   Pexip Infinity will dial out to YouTube. The streaming participant will appear in the participant list with a streaming badge and a name that is typically in the format **a.rtmp.youtube.com**, and a streaming icon is displayed to the right of the layout. (Only one streaming participant is shown, even if you have selected **Dual Stream**.)
2. Wait for a few seconds, then from within your YouTube account, go to your **Live Control Room**.
3. Select **Preview** and confirm.

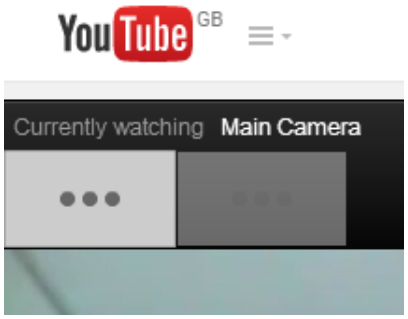


   After a few seconds you will be able to **Play** the Preview stream (of the primary video stream).
   Note that this is your preview only — at this stage the stream is not being broadcast. The stream has a 20-30 second delay.
4. Click **Start streaming** and confirm, to start broadcasting.



You do not have to preview/start the "Camera 2" presentation stream.

5. You are now streaming to anyone who is allowed to access or find your streams (according to your **Privacy** settings). You can optionally **Play** the **Public View** of the stream.
6. You can select **View on Watch Page** (top right of your **Live Control Room** page) to see the normal YouTube view.
   This is how it appears to users who are watching the live stream and is the URL that you should share.
   When you have dual streams, YouTube viewers can switch views between the main camera and the second presentation stream by selecting the thumbnails above the main window.



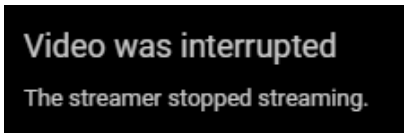
If nobody is currently presenting, Pexip Infinity sends a placeholder image on the presentation stream.

Stopping streaming

This procedure explains how to stop streaming your conference (for either single streamed or dual streamed conferences).

To stop streaming your Pexip conference:

1. Disconnect the streaming participant from the Virtual Meeting Room.
   - When using Infinity Connect: the streaming participant appears in the participant list with a streaming badge and a name that is typically in the format **a.rtmp.youtube.com**.
   - When using the Administrator interface (**Status > Conference**, and then select the conference): the participant alias is the streaming URL with an alias typically in the format **rtmp://a.rtmp.youtube.com.<etc.>**.
2. The Live Control Room will report that the stream status has No Data. and the YouTube public stream will now display:



   Note that while the stream is still open in your YouTube **Live Control Room**, you can restart streaming by adding the RTMP URL to the conference again as a new participant.
3. In your YouTube **Live Control Room**, select **Stop Streaming** and confirm.

4. A few minutes after your stream has ended, it will appear under **Video Manager > Videos**.
   From here you can delete the video if you do not want it to be available for later use, or you can change its privacy settings. You can also use the YouTube video editor to combine multiple recordings, or remove parts of a recording before you publish it etc. (If the video is slow to appear in the **Videos** list, you can also check its content by going to **Video Manager > Live Events** and viewing all ***Completed*** events.)
   Note that if you have produced two recordings (because you selected **Dual Stream**), you must manage each recording separately. When playing back the recordings, YouTube does not provide any mechanism to synchronize them to each other.

Adding a presentation stream when single streaming is already in progress

This procedure explains how to change an existing single streamed RTMP participant into a dual streamed participant.

YouTube ingestion settings cannot be changed after streaming has started. Therefore the only way to add a separate presentation stream after streaming has already started is set up a new, second stream (rather than a second camera on the existing stream) for the presentation channel.

Unlike setting up dual streams from the outset, this time the two streams are completely separate YouTube events and must be viewed in separate browser windows/tabs.

If you have already added a single stream participant to an in-progress conference but want to change this to a dual stream:

1. Go to http://youtube.pexip.com and obtain a second streaming URL. Enter a **Video Name** (e.g. "Alice's training video presentation content" and select **GET url**.
2. From within Infinity Connect, ensure that you have a copy of the existing streaming URL for the video channel:
   - The streaming participant appears in the participant list with a streaming badge and a name that is typically in the format **a.rtmp.youtube.com**.
   - You can select the participant from the participant list and copy the rtmp URL/alias.
3. Disconnect the existing streaming participant from the Virtual Meeting Room.
   Note: do **not** stop streaming in the YouTube **Live Control Room**.
4. Add the streaming participant back in to the conference again, but this time include the second streaming URL:
   a. Select **Add a new participant**.
   b. Select a protocol of ***RTMP*** and a role (we recommend ***Guest***).
   c. Enter the original, existing streaming URL (from step 2).
   d. Select **Dual Stream** and enter the second streaming URL (from step 1).
   e. Select **OK**.
   (Note that administrators can also use the Pexip Infinity Administrator interface to add RTMP streaming participants.)
5. Start broadcasting the second stream in YouTube:
   a. Go to the **Live Control Room** for the second stream.
   (You can select the green link below the rtmp:// address on the Pexip URL generator screen from step 1.)
   b. Select **Preview** and confirm.
   c. **Play** the Preview stream.
   d. Click **Start streaming** and confirm, to start broadcasting the presentation stream.
6. You are now streaming the presentation channel in addition to the original video channel to anyone who is allowed to access or find your streams (according to your **Privacy** settings). You can optionally **Play** the **Public View** of the presentation stream.
7. You can select **View on Watch Page** (top right of your **Live Control Room** page) to see the normal YouTube view and to obtain the URL of the presentation stream that you should share with the conference viewers, alongside the existing video stream URL.

Note that the original conference stream will experience a break in content for the period of time between disconnecting the existing streaming participant from the Virtual Meeting Room and adding it back in again as one of the dual streamed participants. The YouTube **Live Control Room** for the original stream will report that no data is being received, but this will be resolved automatically when the streaming participant is added back into the conference (providing the original streaming URL is used).

Locking a conference and allowing participants to join a locked conference

If you want to prevent any further participants from joining a conference after it has started, you can lock it by using the Administrator interface, using Infinity Connect, or using DTMF. After a conference has been locked, participants who are attempting to join the conference can be allowed in individually by participants already in the conference.

The impact of locking depends on whether or not the Virtual Meeting Room or Virtual Auditorium being used has a Host PIN.

If the service **does not have a Host PIN**:

- Participants will be able to join the conference until it is locked.
- After the conference has been locked, any further participants who attempt to join the conference (including any Automatically Dialed Participants and manually-invited participants) will be held at the **Waiting for conference host** screen.
- All participants who are already in the conference will be notified of any participants who are attempting to join the locked conference, and will be able to allow the waiting participants to join. Notifications will take the form of an on-screen message (visible to Infinity Connect clients only) and a customizable audio message (audible to all participants already in the conference) for each participant attempting to join.
- When the conference is unlocked, any participants who are still waiting will automatically join the conference.

If the service **has a Host PIN**:

- Host and Guest participants will be able to join the conference until it is locked.
- After the conference has been locked, participants who enter the Host PIN will be able to join the conference immediately - locking does not apply to them.
- After the conference has been locked, Guest participants (including any Automatically Dialed Participants and manually-invited participants who have been given a role of Guest) will be held at the **Waiting for conference host screen**.
- All Host participants who are already in the conference will be notified of any Guest participants who are attempting to join the locked conference, and will be able to allow the waiting Guest participants to join. Notifications will take the form of an on-screen message (visible to Infinity Connect clients only) and a customizable audio message (audible to all participants already in the conference) for each participant attempting to join.
- When the conference is unlocked, any Guest participants who are still waiting will automatically join the conference.

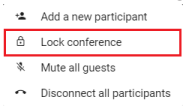
Locking using the Administrator interface

To lock or unlock a conference from the Administrator interface:

1. Log into the Pexip Infinity Administrator interface.
2. Go to **Status > Conferences**.
3. From the **Service name** column, select the conference you want to lock or unlock.
4. At the bottom left of the page, select **Lock conference** or **Unlock conference** as appropriate.

Locking using Infinity Connect

Host participants using Infinity Connect can lock and unlock the conference they are in by clicking on the conference control menu ☰ and selecting **Lock conference** or **Unlock conference** as appropriate:



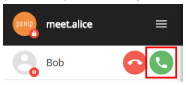
A "locked" icon will appear next to the conference avatar to indicate that the conference is currently locked.

Locking using DTMF

If DTMF controls have been enabled, Host participants can lock and unlock the conference using DTMF. The default DTMF entry to do this is ***7** but this may have been customized. For more information, see Using DTMF to control a conference.

Allowing waiting participants to join a locked conference

When a new participant attempts to join a locked conference, Host participants in the conference who are using Infinity Connect are notified that the participant is waiting to join, and see a red "waiting" icon next to the participant's avatar. To allow the participant to join the locked conference they can click on the green telephone icon next to the participant's name:



In the above example, Bob is waiting to join Alice's locked VMR. Alice is a Host, so can let him join at any time by clicking on the green telephone icon next to Bob's name.

Rejecting a request to join a locked conference

If a Host (who is using Infinity Connect) does not want a waiting participant to join the conference immediately, they have two options:

- To reject the request completely, the Host participant must click on the red telephone icon next to the waiting participant's name. The waiting participant's call will be disconnected.
- To leave the participant at the waiting for Host screen, the Host participant should do nothing. The waiting participant will remain at the waiting screen until:
  - a Host participant chooses to let the waiting participant join the conference, or
  - the conference is unlocked (after which the waiting participant will automatically join the conference), or
  - the conference finishes (after which the waiting participant's call will be disconnected).

Using DTMF to control a conference

Host participants using telephones or SIP/H.323 endpoints that support DTMF can control aspects of the conference by using their keypad.

For information on using Infinity Connect clients to send DTMF tones to another participant, see Using in-call controls.

Default DTMF controls

The default DTMF controls are:

DTMF digits	Control
*7	Toggle conference lock and unlock
*5	Toggle mute and unmute all guests
##	Terminate the conference

Changing DTMF controls

The DTMF digits for each control can be changed on a per-theme basis by editing the themeconfig.json file.

When deciding the digits to be used for each control, you should ensure there is no risk of overlap in situations where one of the digits is not successfully received.

For example, if you used:

- **57** to lock the conference
- **75** to mute all guests
- **77** to end the conference

there is a risk that a user could enter **5775** to lock and mute the conference, but due to packet loss the initial digit was not received. In this case the string that is received would be **775**, which would be interpreted as **77** and thus a command to terminate the conference.

Disabling DTMF controls

DTMF controls are enabled by default. They can be disabled and enabled on a per-theme basis by editing the themeconfig.json file.

Enabling and disabling chat

Conference participants using a variety of clients can chat and share links with other participants who are in the same Virtual Meeting Room or Virtual Auditorium. Supported clients include Microsoft Lync, Skype, and Skype for Business. Chat is also natively

supported in Pexip's own Infinity Connect suite. Microsoft Lync / Skype for Business and Infinity Connect clients can also chat when calling each other directly via the Pexip Distributed Gateway.

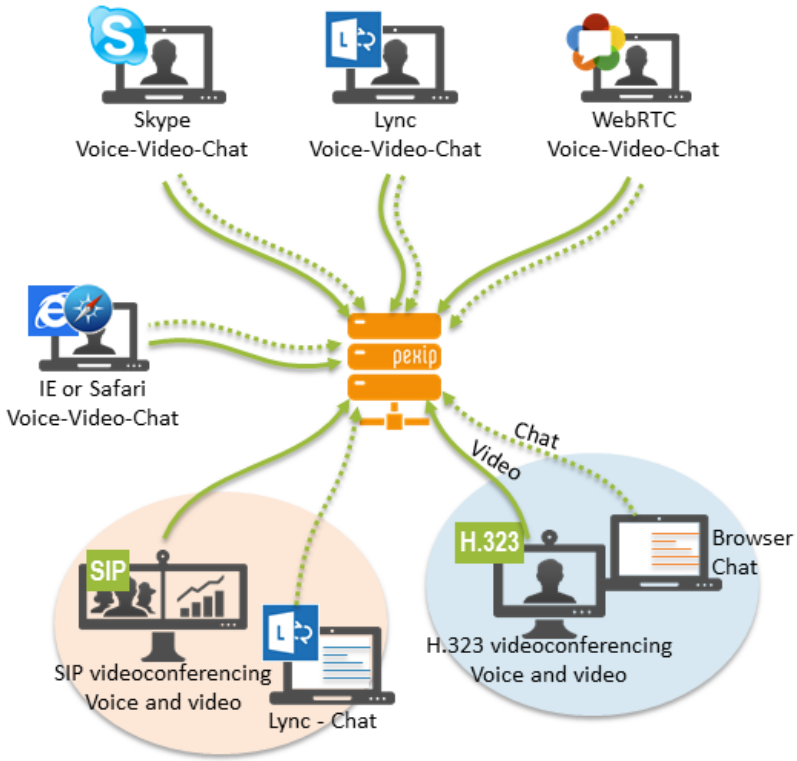
You enable or disable chat on a platform-wide basis, and it is enabled by default. To disable or re-enable this feature:

1. Go to **Platform Configuration > Global Settings**.
2. From within the **Connectivity** section, deselect or select **Enable chat**.

When chat is disabled, Infinity Connect clients will not show the chat window.

Providing chat to participants using unsupported clients

Conference participants who are not using one of the supported clients will not be able to read or send chat messages. However, if they have access to a web browser they can use the Infinity Connect Web App to join the conference without video or audio. This will give them access to the chat room, as well as the ability to view and share presentations, view the participant list, and (if they are Host participants) control aspects of the conference. For more information on using and administering the Infinity Connect suite of clients, see Introduction to Infinity Connect.



Restricting call bandwidth

The Pexip Infinity platform allows you to restrict the amount of bandwidth used by each participant when dialed in to a Pexip Infinity service (Virtual Meeting Room, Virtual Auditorium, Virtual Reception) or for Pexip Distributed Gateway calls or other outbound calls that are managed by Call Routing Rules. You can place restrictions on the bandwidth received by participants, sent by participants, or both. The restriction applies to the total bandwidth of all media (video, audio and presentation) per participant.

Bandwidth restrictions can be configured on a global basis, or on each individual service or routing rule. Settings applied to an individual service override any global limits that have been applied.

If a Virtual Reception has bandwidth limits, these only apply to participants while they are in the Virtual Reception. When their call is transferred to their selected Virtual Meeting Room or Virtual Auditorium, any bandwidth restrictions for that service will then apply.

When sending main video and presentation content to a conference participant, Pexip Infinity always splits the available bandwidth 50/50 between main video and content.

Inbound bandwidth restrictions are implemented as requests to the participant device to limit its bandwidth to the specified amount; Pexip Infinity cannot actually enforce inbound restrictions.

Restricting video resolutions

If you wish to limit video calls to specific resolutions, you should do this using the maximum bandwidth settings, as follows:

To allow resolutions up to...	Set the maximum bandwidth to...
Full HD (1080p)*	no limit
HD (720p)	1800
SD	960

* You must have also enabled Full HD (1080p).

Applying restrictions to an entire deployment

To restrict the bandwidth of calls across your entire deployment:

1. Go to **Platform Configuration > Global Settings**.
2. In the **Service Configuration** section, enter the desired values in the following fields:

Option	Description
Maximum inbound call bandwidth (kbps)	Limits the bitrate of media received by Pexip Infinity from a participant. Leave blank if you do not want to apply any restrictions. Note that this restriction does not apply to specific services or rules that have a **Maximum inbound call bandwidth** configured.
Maximum outbound call bandwidth (kbps)	Limits the bitrate of media sent from Pexip Infinity to a participant. Leave blank if you do not want to apply any restrictions. Note that this restriction does not apply to specific services or rules that have a **Maximum outbound call bandwidth** configured.

Applying restrictions to a service or Call Routing Rule

To restrict the bandwidth of calls to a particular service or Call Routing Rule:

1. Go to the relevant service or rule:
   - **Service Configuration > Virtual Meeting Rooms**
   - **Service Configuration > Virtual Auditoriums**
   - **Service Configuration > Virtual Receptions**
   - **Service Configuration > Call Routing**
2. Either select the name of the service or rule you wish to edit, or click **Add**.
3. In the **Advanced options** section, select **Show** (not required for Call Routing Rules).

4. Enter the desired values in the following fields:

Option	Description
Maximum inbound call bandwidth (kbps)	Limits the bitrate of media received by Pexip Infinity from a participant. Leave blank if you do not want to apply any restrictions. Note that this will override any global **Maximum inbound call bandwidth** that may have been configured.
Maximum outbound call bandwidth (kbps)	Limits the bitrate of media sent from Pexip Infinity to a participant. Leave blank if you do not want to apply any restrictions. Note that this will override any global **Maximum outbound call bandwidth** that may have been configured.

5. Select **Save**.

Examples

- If you want to limit a particular Virtual Meeting Room to standard-definition (SD) video calls, you should restrict the inbound and outbound call bandwidth to 960 kbps — this allows for the highest quality SD but ensures that HD is not available.
- If you have enabled Full HD (1080p) in your deployment, but wish to limit some Virtual Meeting Rooms to HD (720p) calls only, you should restrict the inbound and outbound call bandwidths of that service to 1800 kbps.
- If you want to allow certain Virtual Meeting Rooms to support HD (720p) calls, but restrict all other services to SD calls, then you should apply:
  - maximum inbound and outbound bandwidths of 960 kbps globally, and
  - maximum inbound and outbound bandwidths of 1800 kbps to the Virtual Meeting Rooms that you want to be able to use HD.
- In most cases you should apply the same bandwidth restrictions to the inbound and outbound calls within a service. However, you may wish to allow participants to send their video as SD, but receive the composed layout of all participants (main video and video thumbnails) as HD. In this case, you would set the inbound call bandwidth to 960 kbps and the outbound to 1800 kbps.

Limiting the number of participants

You can place a limit on the number of participants that can access a particular Virtual Meeting Room or Virtual Auditorium at any one time. When such a restriction is in place, after the limit has been reached any new participants will be presented with a **capacity exceeded** message.

If you change this setting while a conference is in place, participants who have already joined the conference will not be affected, even if the conference is already over capacity. However, after a minute or so (when the change has been replicated to all Conferencing Nodes) any new participants attempting to join the conference will be rejected.

Infinity Connect users who have joined as presentation and control only participants will count towards the participant limits.

Presentation streams from standards-based endpoints and Lync / Skype for Business clients are not counted separately; such participants are only counted once regardless of whether or not there is a presentation stream in addition to the main video.

If you do not restrict the number of participants, any number of participants will be able to join the service, subject to the available capacity of your Pexip Infinity deployment.

To limit the number of participants in a particular Virtual Meeting Room or Virtual Auditorium:

1. Go to **Service Configuration > Virtual Meeting Rooms** or **Service Configuration > Virtual Auditoriums**.
2. Select the Virtual Meeting Room or Virtual Auditorium to change.
3. In the **Advanced options** section, select **Show**.
4. In the **Participant limit** field, enter the maximum number of participants you wish to be able to use the service at any one time.
5. Select **Save**.

Controlling media capability

Calls consist of three types of media - audio, video, and presentation - each in their own stream. Pexip Infinity allows you to control which types of media can be used for certain calls. This feature allows better resource management and smaller SDPs when calling out to known audio-only devices.

Limiting media capability

You can restrict the media capability either on a per-call basis (the call capability) or a per-conference basis (the conference capability), as follows:

Adding a participant to a conference

When dialing out to a participant from within a conference, either manually or automatically, you can choose whether the call will be **audio-only**, **main video only**, or **main video plus presentation**. In such cases, the called participant will not be able to escalate the call (for example, from an audio-only call to video).

Conference-wide limitations

You can apply limits to individual Virtual Meeting Rooms or Virtual Auditoriums, restricting them to be **audio-only**, **main video only**, or **main video plus presentation**. In such cases, participants calling in to the conference with a higher media capability will have their media limited. For example, when a participant dials in over video to an audio-only VMR, their call will be placed as audio-only and they will not subsequently be able to escalate the call.

Using the **audio-only** setting where appropriate also has the advantage of increasing the overall capacity of your distributed deployment. Each Conferencing Node will reserve an extra connection for each Virtual Meeting Rooms and Virtual Auditoriums it is hosting, to be used for a backplane should that service become distributed (i.e. hosted on more than one Conferencing Node). By default this connection will be equivalent in capacity to an HD call, but if the service has been configured as **audio-only** it will instead be equivalent to that of an audio-only call. For further information on capacity and how calls consume resources, see Hardware resource allocation rules.

Virtual Receptions

You can limit the conference capability of a Virtual Reception. Any restrictions will be applied to a call while it is accessing the Virtual Reception service. When the call is placed to the destination Virtual Meeting Room or Virtual Auditorium, the capability of that service will be applied instead. (Note however that some endpoints may maintain the restriction when the call is transferred to the destination conference.)

For example, if you limit your Virtual Reception to audio-only, when Alice calls it she will not receive any video and will only hear the audio prompts. However, when her call is then placed to the destination VMR which has a capability of **main video**, she will then be able to send and receive video as well as audio.

Pexip Distributed Gateway calls

For calls made using the Distributed Gateway, you can configure the relevant Call Routing Rule to limit the media by selecting an appropriate **Call capability** setting:

- You can limit the media capability of the outbound call to **audio-only**, **main video only**, or **main video plus presentation**. For example, if the media capability is set to **audio-only** and Alice makes a video call to Bob, Bob will not be able to answer with video.
- You can match the capability of the outbound call to that of the inbound call by using the **Same as incoming call** option. This means that if, for example, Alice makes an audio-only call to Bob, Bob will not be able to answer with video.

  Note that, when using **Same as incoming call**:
  - an audio-only call cannot later be escalated to video
  - auto-escalation of Lync / Skype for Business calls (if enabled) will not work

Automatically escalating Lync / Skype for Business audio calls

You can control how Pexip Infinity handles incoming audio-only calls from Lync / Skype for Business* clients.

By default, Pexip Infinity treats the call as audio-only and does not send video back to the Lync/SfB participant.

This behavior can be changed on a platform-wide basis by selecting the **Enable Lync / Skype for Business auto-escalation** setting on the **Platform Settings > Global Configuration** page.

When Lync/SfB auto-escalation is enabled, Pexip Infinity automatically escalates a Lync/SfB audio-only call so that it receives video from a conference (the Lync/SfB participant still only sends audio).

The following table shows the differences in conference experience based on the auto-escalation setting when a Lync/SfB client makes an audio call to a Pexip conference:

	Lync/SfB auto-escalation configuration	
	Disabled (default)	**Enabled**
The Lync/SfB participant...	sees the conference avatar image (pexip by default)	receives a video stream from the conference
Other conference participants see the Lync/SfB participant as...	an audio-only participant indicator on the left side of the video window (by default)	a "no incoming video" indicator when the participant is shown as the main speaker or as a thumbnail (by default)

In all cases, the Lync/SfB participant can still subsequently escalate to send (and receive) video.

* Note that where this documentation refers to "Lync/SfB", it represents both Microsoft Lync and Skype for Business unless explicitly stated otherwise.

Automatically ending a conference

You may want to ensure that conferences are automatically terminated in certain situations, in order to preserve resources and to ensure that certain types of participants can't keep the conference alive by remaining in a call indefinitely. For this reason, there are settings that allow you to end a conference in any of the following situations:

- when only Guests remain
- when there is only one participant remaining in the conference
- when the only participants remaining are ADPs or participants that have been added by an administrator.

These settings are all independent; a conference needs to meet only one of the above criteria in order for it to be terminated automatically on that basis. For example, you may have configured your system to terminate conferences 120 seconds after the last Host leaves, and 60 seconds after there is only one participant remaining. In that case:

- if all but one Guest remains in a conference 30 seconds after the last Hosts has left, the conference will be terminated 60 seconds later (i.e. based on the one participant remaining setting)
- if all but two guests remain 30 seconds after the last Host has left, the conference will be terminated in another 90 seconds (i.e. based on the last Host setting).

When only guests remain

Pexip Infinity will always terminate a conference a certain period after the last Host* disconnects and only Guests remain in the call. By default this period is set to one minute, but it can be set to between 0 seconds (i.e. all Guests are disconnected immediately upon the last Host disconnecting), and one day.

The default is 60 seconds. This should be sufficient time for a Host who has been unintentionally disconnected from a conference to reconnect before the conference is terminated.

* For the purposes of this setting, in order to keep the conference alive a Host must have either joined with audio or video, or joined as control-only and elected to Start conference. Other control-only Hosts will not keep the conference alive.

To change this setting, go to **Platform Configuration > Global Settings > Guests-only Timeout**.

When there is only one participant remaining in the conference

You can configure the length of time (in seconds) for which a conference will continue with only one participant remaining. This can be set to between 60 seconds and one day, or alternatively you can set it to ***0*** which means that a conference will never be

terminated on the basis that there is a single participant remaining. The default is ***0***.

When this setting is used to terminate a conference, the type of participant (Host, Guest, ADP, administrator-added, streaming, etc.) is irrelevant; if they are the only participant remaining, the conference will be terminated after the specified time.

To change this setting, go to **Platform Configuration > Global Settings > Last Participant Backstop Timeout**.

When the only participants remaining are ADPs and/or administrator-added

In most cases a conference won't be terminated if a Host participant is present. However, if the only remaining Host participants are:

- Automatically Dialed Participants (ADPs), and/or
- participants who have been added to a conference by an administrator (either from the Administrator interface or using the Management API),

then the conference may or may not be terminated, depending on the participants' **Keep conference alive** settings and whether any other ADPs or administrator-added participants remain in the conference.

Keeping a conference alive

There are three options for the **Keep conference alive** setting:

- ***Yes***: if the participant is a Host, the conference will continue to run even when this is the only participant remaining - in other words, they are just like any other Host participant. This is the default used when adding a participant to a conference using the Administrator interface or using the Management API.
- ***No***: the conference will be automatically terminated if this is the only participant remaining. This is the option we recommend for automated systems that are unable to terminate a call themselves, such as streaming participants.
- ***If multiple***: if two or more ADP or administrator-added participants with the ***If multiple*** setting remain in the call and at least one of them is a Host, the conference will not be terminated by Pexip Infinity. However, if all other participants have disconnected and only one ADP or administrator-added participant with the ***If multiple*** setting remains, the conference will be terminated by Pexip Infinity. This is to prevent automated systems (such as recording devices) that are unable to terminate a call themselves from keeping the conference alive indefinitely. For this reason, if you are using this option we recommend that each Virtual Meeting Room or Virtual Auditorium has no more than one such automated system as an ADP. A better alternative is to give automated systems a **Keep conference alive** setting of ***No*** (see above).

  If multiple is the default option when adding an ADP.

If the only remaining participants in a conference are ADPs and/or administrator-dialed participants with a mix of ***Yes***, ***If multiple*** and ***No*** options, the conference will be terminated unless:

- at least one participant has a **Keep conference alive** setting of ***Yes***, or
- there are two or more participants with a setting of ***If multiple*** and at least one of them is a Host.

Muting a participant's audio

There are several ways the audio being sent from a participant can be muted:

- Using the Administrator interface
- Using Infinity Connect
- Using the Infinity Connect Mobile client for iOS
- Using DTMF
- The Pexip API can also be implemented by third party applications to mute a participant. For more information, see Introduction to the management API.

When a participant's audio has been muted, Pexip Infinity will still receive their audio stream but will not add it to the mix being sent to all other participants.

Using the Administrator interface

To use the Pexip Infinity Administrator interface to mute a participant's audio:

1. Select the participant. You can do this in two ways:
   - Go to **Status > Participants** and select the participant to mute.
   - Go to **Status > Conferences** and select the Virtual Meeting Room or Virtual Auditorium that the participant is in. From the **Participants** tab, select the participant to mute.
2. At the bottom right of the screen, select **Mute**.

Using Infinity Connect

You must have Host privileges to use this feature.

It is possible to mute a participant's audio using Infinity Connect. Note that this will not mute the participant's speakers, so they will still hear all other unmuted participants, but what they say will not be heard. Host participants can mute an individual participant, or mute all Guests simultaneously.

When Infinity Connect has been used to mute a participant, the **Audio administratively muted?** column of the Conference status page of the Pexip Infinity Administrator interface will show ***YES***.

Participants will not be aware that they have been muted or unmuted.

Participants can mute and unmute themselves using Infinity Connect, but only if they have Host privileges.

An Infinity Connect user can unmute a participant previously muted by another Infinity Connect user.

Muting an individual participant

To use Infinity Connect to mute an individual participant's audio:

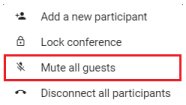
1. From the participant list, select the **Mute** icon to the right of the participant's name.

   A "muted" icon will appear next to the participant's avatar, and the icon will appear next to their name.
2. To unmute the participant, select the **Unmute** icon .

Muting all Guest participants

To use Infinity Connect to mute the audio coming from all Guests participants:

- From the top left of the screen, select the menu icon and then select **Mute all guests**:



A "muted" icon will appear next to the conference avatar. All Guest participants currently in the conference, and any Guest participants who subsequently join the conference, will be muted (indicated by a "muted" icon next to their avatar).

Using the Infinity Connect Mobile client for iOS

To mute or unmute a conference participant's audio:

1. Select the participant.
2. Toggle the **Enable audio from participant** setting:

Using DTMF

If DTMF controls have been enabled, Host participants can mute and unmute all Guest participants. The default DTMF entry to do this is ***5** but this may have been customized. For more information, see Using DTMF to control a conference.

Disconnecting participants from a conference

There are several ways to disconnect a participant from a conference: Using the Administrator interface, Using Infinity Connect, Using the Infinity Connect Mobile client for iOS and Using DTMF. Each of these options is described below.

Third-party applications can also use the management API to disconnect participants from a conference. For more information, see Introduction to the management API.

Using the Administrator interface

Disconnecting a single participant

To use the Pexip Infinity Administrator interface to disconnect a single participant from a conference:

1. Select the participant. You can do this in two ways:
   - Go to **Status > Participants** and select the participant to disconnect.
   - Go to **Status > Conferences** and select the Virtual Meeting Room or Virtual Auditorium that the participant is in. From the **Participants** tab, select the participant to disconnect.
2. At the bottom right of the screen, select **Disconnect**.

Disconnecting all participants from a conference

To use the Pexip Infinity Administrator interface to disconnect all of the participants from a conference:

1. Go to **Status > Conferences** and select the Virtual Meeting Room or Virtual Auditorium that the participants are in.
2. At the bottom right of the screen, select **Disconnect all**.
3. Select **Yes, I'm sure** to confirm.

All participants will be disconnected and the conference will no longer be active.

Using Infinity Connect

You must be a Host participant in the conference to use this feature.

Disconnecting a single participant

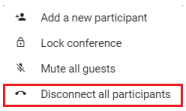
To use an Infinity Connect client to disconnect a participant:

- From the participant list, select the **Disconnect** icon to the right of the participant's name.

Disconnecting all participants

To use an Infinity Connect client to disconnect all participants, including yourself:

- From the top left of the screen, select the menu icon and then select **Disconnect all participants**:



Using the Infinity Connect Mobile client for iOS

You must have Host privileges to use this feature.

To remove a participant from the conference:

1. Select the participant, or left-swipe the participant.
2. Select **Disconnect participant**.

Using DTMF

If DTMF controls have been enabled, Host participants can terminate the conference by disconnecting all participants, including themselves. The default DTMF entry to do this is ## but this may have been customized. For more information, see Using DTMF to control a conference.

Customizing with themes

Customizing images and voice prompts using themes

Themes allow you to change the voice prompts and images provided to participants when they are accessing a Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service, or making a person-to-person call using the Distributed Gateway. You might change the theme if, for example, you want to use your company's own logo or color scheme on the PIN entry screens, or you want to change the language used in the voice prompts.

Each conference theme is made up of a customized set of audio and image files. You can replace any or all of these files with your own customized sounds and images in order to create a new theme. If required, you can automatically generate the images used for the PIN entry screens based on a background image and color of your choosing.

You can also use themes to apply a video watermark to your conferences.

You apply themes to individual Virtual Meeting Rooms, Virtual Auditoriums, Virtual Receptions, Test Call Services and Call Routing Rules, but the same theme can be used for multiple services. You can also nominate a global default theme to be used for any services that have not had a specific theme applied to them.

Pexip Infinity ships with its own Base theme, which cannot be edited. Files from the Base theme are used when no other theme has been selected, or when the selected theme does not contain a specific file. Pexip Infinity also ships with a number of preconfigured themes — these include alternative themes that contain entry and exit tones and localized audio files — making it easier for you to select and create themes suitable for your deployment.

Note that if you use the Infinity Connect clients to access your services, you can also customize the look and feel of those clients. For more information, see Customizing the Infinity Connect Web App.

How do I know which files will be used?

When you create a new customized theme, you don't have to upload a complete set of files — you can just upload the ones you want to be used specifically for that theme. When a user accesses a service that has that theme applied, Pexip Infinity presents them with the sounds and images from the files included in that theme. If the customized theme does not include a specific file, or the service does not have a theme applied, then Pexip Infinity uses the relevant file from the theme that has been selected as the default theme. If the default theme does not include the specific file either, then the file from the Base theme is used.

Example - replacing the logo in the PIN entry screen

Imagine you want to replace the Pexip logo that appears in all the PIN entry screens with your own company's logo, and have this apply to every VMR in your deployment. To do this:

1. Create a new customized "Logo" theme containing only the new PIN entry screens.
2. Select this new theme as your default theme.

Now, users will see your own company's logo when entering their PIN, but all other image files, and all the audio prompts, would come from the Base theme.

Example - using individual avatars for Lync / Skype for Business contacts

Imagine you want to change the avatar that is displayed in Lync / Skype for Business users' contact lists to represent Alice's and Bob's personal VMRs, so that they show Alice's and Bob's images respectively. To do this, you create a new customized "Alice" theme that contains only Alice's image (saved as **presence_avatar_image.jpg**) and a new customized "Bob" theme that contains only Bob's image (also saved as **presence_avatar_image.jpg**). You then select the "Alice" theme for Alice's personal VMR, and the "Bob" theme for Bob's personal VMR. Now, Lync / Skype for Business users will see Alice's and Bob's images as the avatars representing their respective VMRs.

Example - replacing the logo and using avatars together

Imagine you want to use individual avatars for each VMR and also replace the Pexip logo with your corporate logo. To do this, you would combine the steps in the above two examples - apply a theme containing just the avatar image file to each VMR, and then select the theme that uses the corporate logo as your default theme. Now, Lync / Skype for Business users will see Alice's and Bob's

images as the avatars representing their personal VMRs, and when they connect to the VMR they will see your company's logo in the PIN entry screens. All other image files, and all the audio prompts, will come from the Base theme.

Viewing the themes currently in use

To find out which themes are currently in use in your deployment, go to **Service Configuration > Themes**. The **Service count** column will show you how many services each theme is associated with.

Themes are also shown on the overview page for each type of service (go to **Service Configuration** and then select the service). Here you can sort the list by **Theme**.

Creating and editing themes

This section describes how to create and edit your own themes, including how to automatically generate the PIN entry screen images, and how to change which themes are associated with which services.

Creating a new theme

You can create a new theme by uploading a .ZIP file containing the files you want to use instead of the default theme.

The .ZIP file does not need to contain the complete set of files, just any files that you want to use in place of those from the default theme.

To create a new theme:

1. Ensure that all new sound and image files meet the specified file requirements for themes.
2. Save all new sound and image files together in a new folder. You can name the folder whatever you like, but within the folder, each new file must be saved with the same file name and extension as the default file being replaced.
3. Create a .ZIP file of the new folder.
4. Go to **Service Configuration > Themes** and select **Add theme**.
5. In the **Name** field, enter the name of the new theme.
6. In the **Theme** field, select **Choose file** and go to the .ZIP file you have just created.
7. In the **Services** and **Call Routing Rule** sections, select the services (Virtual Meeting Rooms, Virtual Receptions, Call Routing Rules and so on) that will use this new theme.
   Note that if you have more than 10,000 services configured, you cannot select the associated services via this page. Instead you must use another method to assign the theme to a service, such as editing the individual service, using the configuration API, or by using the bulk VMR provisioning facility.
8. Select **Save**.
9. Allow a minute or so for the new theme to be replicated out to all Conferencing Nodes.

You can now test the theme by dialing one of the services to which it has been applied.

Editing an existing theme

You can edit an existing theme by uploading a new .ZIP file containing the new set of files.

The .ZIP file does not need to contain the complete set of files, just any files that you want to use in place of those from the default theme.

Note that when you edit a theme, all existing files in that theme will be deleted and replaced with files from the new theme. This means that if you have an existing theme containing **File A** and you want to add **File B** to the theme, you need to upload a .ZIP file containing both **File A** and **File B**.

To edit an existing theme:

1. If you want to retain files from the existing theme, first download the theme.
2. Ensure that all new sound and image files meet the specified file requirements for themes.
3. Save all new sound and image files, along with any existing files, together in a new folder. Each file must be saved with the same file name and extension as the default file being replaced.

4. Create a .ZIP file of the new folder.
5. Go to **Service Configuration > Themes** and select the theme you wish to edit.
6. In the **Theme** field, select **Choose file** and go to the .ZIP file you have just created.
7. Select **Save**.
8. Allow a minute or so for the updated theme to be replicated out to all Conferencing Nodes.

You can now test the theme by dialing one of the services to which it has been applied.

Downloading an existing theme

You can download a theme to view the files it contains, or to edit it.

To download a theme:

1. Go to **Service Configuration > Themes**.
2. Either:
   - to download a copy of the default set of files that are shipped with Pexip Infinity, select **Download Base theme**
   - to download a copy of the files that make up a customized theme, select the theme and from the **Change Theme** page select **Download theme**.

A file with a .ZIP extension will be downloaded.

Automatically generating the PIN entry screen images

To save you from having to create 23 separate images for each of the PIN entry screens (i.e. all Image files with a file name beginning with **pin_entry_**), you can provide a background image and foreground color, and then allow Pexip Infinity to automatically generate the required images for you.

You could use this option if you want to use the new images as part of an entirely new theme, or if you want to edit the PIN entry screens only.

The background image must be:

- JPEG format
- 1920x1080 pixels.

The PIN entry indicators will:

- begin 634 pixels from the top of the background image
- be 70 pixels in diameter
- have 14 pixels of space between each indicator.

You must ensure that the PIN entry indicators use a color that makes them clearly visible when overlaid on the background image.

To change the PIN entry screen images:

1. Go to **Service Configuration > Themes**.
2. Select **Generate theme**.
3. In the **Background image** field, select **Choose file** and select the image you want to appear behind the PIN entry indicators.
4. In the **Foreground color** field, enter the hexadecimal value of the color you want to use for the PIN entry indicators.
5. Select **Generate**.
6. A <image name>.ZIP file will be downloaded. This file will contain the 23 new PIN entry images.
7. If you want to add any other new files to this theme:
   a. Unzip the downloaded .ZIP file to a new folder.
   b. Add the new files to the new folder.
   c. Create a .ZIP file of the new folder.
8. Go back to **Service Configuration > Themes** and select **Add Theme**.
9. In the **Name** field, enter the name you want to give to this new theme.

10. In the **Theme** field, select **Choose file** and go to the .ZIP file you have just created.
11. In the **Services** and **Call Routing Rule** sections, select the services and rules that will use this new theme.
12. Select **Save**.
13. Allow a minute or so for the new theme to be replicated out to all Conferencing Nodes.

You can now test the theme by dialing in to one of the VMRs to which it has been applied.

Changing which themes are associated with which services

There are many ways in which you can change the theme that is associated with a service or Call Routing Rule). You can do this:

- via the **Themes** page (go to **Service Configuration > Themes**, select the new theme you want to use, then select the services you want to apply it to) — you cannot use this method if you have more than 10,000 services configured
- by editing the individual service (go to **Service Configuration**, select the service, then select the new theme from the **Theme** drop-down menu) or Call Routing Rule (**Service Configuration > Call Routing**)
- via the configuration API
- by using the bulk VMR provisioning facility.

Setting the default theme

You can specify a theme to be used by default for all services that have not had a specific theme applied.

The default theme does not need to contain a complete set of files, just those that you want to be used in place of files from the Base theme.

If you do not select a default theme, the entire Base theme will be used as the default.

To change the default theme:

1. Go to **Platform Configuration > Global Settings**.
2. From the **Default theme** drop-down, select the theme to be used for services where no specific theme has been selected. If you have not already uploaded the theme you wish to use, you can do so from here by clicking on the ✚ icon and following the instructions in Creating and editing themes.
3. Select **Save**.

File requirements for themes

The following sections list the complete set of audio, image and configuration files that make up a full theme, including how to manage video watermarking. This is also the same set of files that comprise the Base theme. When creating or editing a theme, you upload your own customized version of one or more of these files, ensuring each new file name is identical to that of the file being replaced.

All files must meet the requirements specified at the start of each section.

Audio

All sounds must be:

- .WAV format
- RIFF (little-endian) data, WAVE audio, Microsoft PCM, 16 bit, mono 48000 Hz

Points to note:

- **mono** and **48000Hz** are essential - audio files that do not meet these requirements will fail to upload.
- The volume level of the audio recording is important - use the default Pexip Infinity prompts as a guide.
- Some endpoints may take a few seconds after a call connects before they are able to receive audio. For this reason, we have included a 2-second pause at the start of any audio files that may be played when a user first connects to Pexip Infinity. We recommend that you include a similar pause; use the default Pexip Infinity files as a guide.

Audio files

File name	Content in Base theme
2sd-invalid-number-three-times-disconnect_ 48kHz_mono.wav	"You have entered an invalid number three times. I will now disconnect the call."
2sd-not-entered-valid-number-disconnect-call_ 48kHz_mono.wav	"You have not entered a valid number. I will now disconnect the call."
2sd-number-not-valid-try-again_48kHz_mono.wav	"That number is not valid. Please try again."
2sd-number-pound-key_48kHz_mono.wav	"Please enter the number you wish to connect to, followed by the pound key." †
2sd-please-hold-connect-you_48kHz_mono.wav	"Please hold while I try to connect you."
conf-call-will-be-disconnected_48kHz_mono.wav	"Your call will be disconnected."
conf-capacity_exceeded_48kHz_mono.wav	"The conferencing system capacity has been exceeded."
conf-getpin_48kHz_mono.wav	"Please enter the conference PIN number."
conf-getpin_pound-key_48kHz_mono.wav	"Please enter the conference PIN number, followed by the pound key." †
conf-insufficient_licenses_48kHz_mono.wav	"There are insufficient conferencing system licenses available."
conf-invalid_license_48kHz_mono.wav	"The conferencing system license is invalid."
conf-invalidpin_48kHz_mono.wav	"The PIN is invalid for this conference."
conf-leaderhasleft_48kHz_mono.wav	"The Host has left the conference. The conference is about to end."
conf-participant_entry_tone_48kHz_mono.wav	<empty file> (For more information, see Playing notification tones when participants join or leave a conference.)
conf-participant_exit_tone_48kHz_mono.wav	<empty file>
conf-participant_locked_out_48Khz_mono.wav	Three knocks. (For more information, see Locking a conference and allowing participants to join a locked conference.)
conf-placeintoconf_48kHz_mono.wav	"Welcome to the conference."
conf-test_call_48kHz_mono.wav *	"Let's test your video and audio. Count out loud from one to three, now." (For more information, see Configuring the Test Call Service.)
conf-test_call_disconnect_48kHz_mono.wav *	"If you have technical issues, check your settings or contact your administrator."
conf-waitforleader_48kHz_mono.wav	"Waiting for the conference Host to join."
conf-waithostpin_48kHz_mono.wav	"Waiting for the conference Host to join. If you are the conference Host, please enter the conference PIN number now."
conf-waithostpin_pound-key_48kHz_mono.wav	"Waiting for the conference Host to join. If you are the conference Host, please enter the conference PIN number followed by the pound key." †
conf-you_are_the_only_participant_48kHz_ mono.wav	"You are the only participant in the conference."

* Added in v14.

† Alternative recordings using the term "hash key" are also available - see Preconfigured themes.

Images

Each image must be a specific size and file format (PNG or JPEG), as specified in Image files. PNG images support transparency; JPEG images do not.

Points to note:

- Ensure that the images display acceptably on both 16:9 endpoints and 4:3 endpoints (if applicable). For example, the original 1920 x 1080 pixel (16:9) images may have approximately 240 pixels cropped from both the left and right sides when displayed as 4:3. You should therefore check that any details on the far left or far right of the image are visible in both formats; for this reason we recommend keeping the image fairly central.
- Depending on the endpoint capabilities and network conditions, images may be scaled down to CIF (352 x 288) or even QCIF (176 x 144), so ensure that images are not overly detailed, and any text uses a clear, large font to ensure that it is legible at low resolutions.
- The transfer and two_stage_dialing images will be overlaid with system-generated text. By default this text is white, but the color can be changed via the themeconfig.json file. When creating customized themes, you must therefore ensure these images include sufficient empty space starting from 755 pixels from the top of the image, in a contrasting color to the text, to allow the text to be visible.
- The default font for the overlaid system-generated text is Roboto, or if that is not available for the selected language, Noto Sans. Icons are from the Material Design icon library.
- Pexip provides some resources for constructing your own themes at http://download.pexip.com/resources/themes/. This includes white and black colored versions of the "no other participants" and "no other video participants" graphics. They are provided as transparent PNG files and can be used as overlays on top of your own background JPEG images.

Video watermarking

Video watermarking overlays a small transparent image onto the main speaker video during a conference (VMRs and Virtual Auditoriums). Watermarks are not added to calls placed via the Pexip Distributed Gateway.

The default watermark is a white "Pexip" logo. You can change the watermark image or disable watermarking completely.

When watermarking is enabled:

- The contents of the **watermark_icon.png** file is used as the watermark image.
- The top-left corner of the watermark is placed 56 pixels across and 40 pixels down from the top-left corner of the main speaker video, and the watermark area then extends 200 pixels across and 100 pixels down. These sizes are based on Full HD resolution — they will be smaller on lower resolutions.
- The size and position of the watermark area cannot be customized.

Enabling or disabling watermarking

The **disable_watermark_icon** setting in the themeconfig.json file controls whether watermarking is enabled or not:

- Set **disable_watermark_icon** to ***false*** to enable watermarking.
- Set **disable_watermark_icon** to ***true*** to disable watermarking.

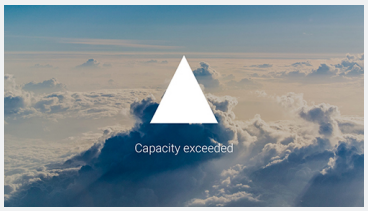
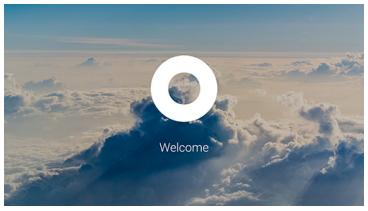
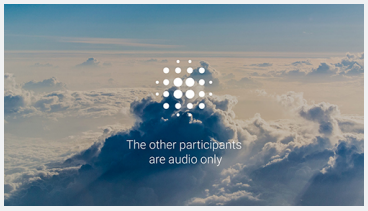
By default, **disable_watermark_icon** is set to ***false***.

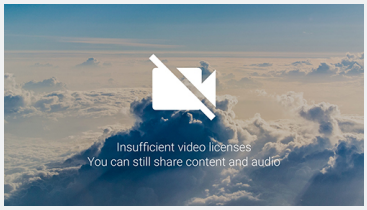
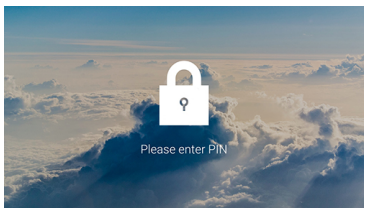
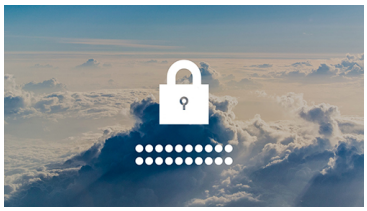
Changing the watermark image

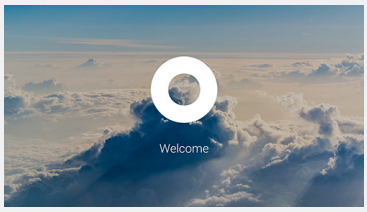
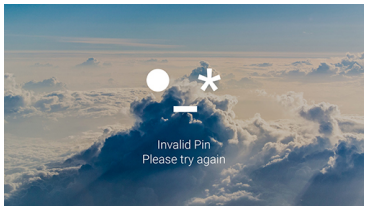
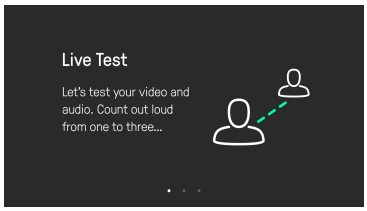
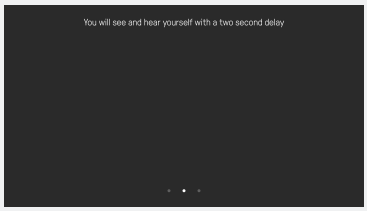
When creating your own watermark file (**watermark_icon.png**) for inclusion in a theme:

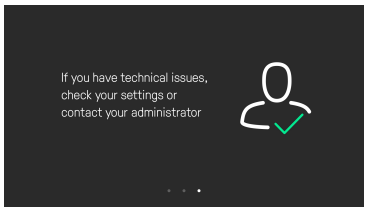
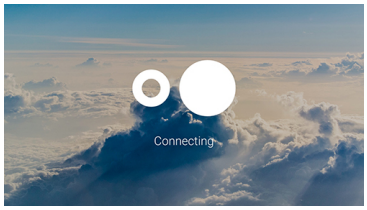
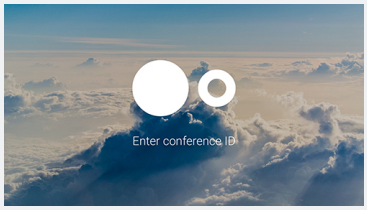
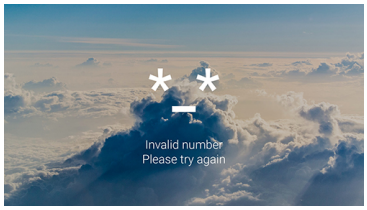
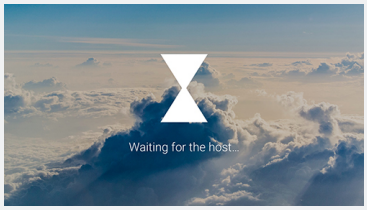
- The file must be PNG format and include a transparent image (if it is not transparent the overlaid content will be fully opaque).
- We recommend that the watermark image is white with 40% transparency.
- The image should be 200 pixels wide x 100 pixels high. Larger images are automatically scaled to fit to the top left corner of the watermark area (and thus may lose some definition).

Image files

File name	Image size (width, height)	File format	Content in Base theme	Notes
audio_indicator.png	100 x 100 pixels	.PNG		Used to represent an audio-only participant.
capacity_exceeded_image.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Shown to participants (other than those using Infinity Connect) when they attempt to join a conference that has reached its maximum number of participants. For more information, see Limiting the number of participants.
insufficient_license_image.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Shown to participants (other than those using Infinity Connect) when they are unable to join a conference because all call licenses are currently in use. For more information, see Insufficient licenses.
invalid_license_image.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Shown to participants (other than those using Infinity Connect) when they are unable to join a conference because the Pexip Infinity license is not valid. For more information, see Invalid license.
no_incoming_video_icon.png	400 x 400 pixels	.PNG		Shown when a participant's video stream is not available.
no_main_video_icon.png	400 x 400 pixels	.PNG		Used when streaming a conference to indicate that there is currently no main video.
no_other_participants.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Shown when there are no other participants in the conference.
no_other_video_participants.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Shown when all other participants are audio-only or control-only.

File name	Image size (width, height)	File format	Content in Base theme	Notes
no_presentation_video_icon.png	400 x 400 pixels	.PNG		Used when dual streaming a conference to indicate that there is currently no presentation.
no_video_license.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Shown to participants when they are unable to join a conference because all port (video) licenses are currently in use, but there was an audio license available. For more information, see Insufficient licenses.
pin_entry_0_digits_image.jpg ‡	1920 x 1080 pixels	.JPG (RGB mode only)		Shown to participants (other than those using Infinity Connect) when entering PINs.
pin_entry_1_digits_image.jpg ‡	1920 x 1080 pixels	.JPG (RGB mode only)		
pin_entry_2_digits_image.jpg ‡	1920 x 1080 pixels	.JPG (RGB mode only)		
pin_entry_3_digits_image.jpg ‡ ... and so on, up to...	1920 x 1080 pixels	.JPG (RGB mode only)	 ↓	
pin_entry_20_digits_image.jpg ‡	1920 x 1080 pixels	.JPG (RGB mode only)		

File name	Image size (width, height)	File format	Content in Base theme	Notes
pin_entry_correct_image.jpg ‡	1920 x 1080 pixels	.JPG (RGB mode only)		
pin_entry_incorrect_image.jpg ‡	1920 x 1080 pixels	.JPG (RGB mode only)		
plus_n_indicator.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Background of the thumbnail on which the text showing the number of participants is overlaid. Note: the color of the text is specified in the themeconfig.json file.
presence_avatar_image.jpg	128 x 128 pixels	.JPG (RGB mode only)		Shown next to the conference alias when using Infinity Connect clients, and as the contact avatar when using Lync / Skype for Business. When the image is used, the corners will be cropped in order to be displayed as a round image.
streaming_indicator.png	100 x 100 pixels	.PNG		Used to indicate that the conference is being streamed or recorded.
test_call.jpg *	1920 x 1080 pixels	.JPG (RGB mode only)		Shown at the start of a call to a Test Call Service. (For more information, see Configuring the Test Call Service.)
test_call_during_recording.jpg *	1920 x 1080 pixels	.JPG (RGB mode only)		Shown during a call to a Test Call Service. Note that a large, live (with a short delay) video image of the test call participant is shown on top of this JPG file during a test call.

File name	Image size (width, height)	File format	Content in Base theme	Notes
test_call_disconnect.jpg *	1920 x 1080 pixels	.JPG (RGB mode only)		Shown briefly prior to automatically disconnecting the participant from a Test Call Service.
timed_out.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Shown briefly prior to disconnecting participants (other than those using Infinity Connect) who have connected to a Virtual Reception but did not enter a valid conference number.
transfer.jpg †	1920 x 1080 pixels	.JPG (RGB mode only)		Shown briefly to participants (other than those using Infinity Connect) while being transferred from a Virtual Reception to a Virtual Meeting Room or Virtual Auditorium.
two_stage_dialing.jpg †	1920 x 1080 pixels	.JPG (RGB mode only)		Shown to participants (other than those using Infinity Connect) who have connected to a Virtual Reception.
two_stage_dialing_ conf_number_invalid_ image.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Shown to participants (other than those using Infinity Connect) who have connected to a Virtual Reception and have entered an invalid conference number.
waiting_for_ conference_leader_ image.jpg	1920 x 1080 pixels	.JPG (RGB mode only)		Shown to Guests while waiting for a Host to join.

File name	Image size (width, height)	File format	Content in Base theme	Notes
watermark_icon.png	200 x 100 pixels	.PNG		Transparent image used for applying a watermark to the main speaker video in a conference. The default image is a white Pexip logo with 40% transparency and is shown here against a blue background. See Video watermarking for more information.

‡ When creating customized themes, these files can be automatically generated using a background image and foreground color of your choice.

† These images will be overlaid with system-generated text. By default this text is white, but the color can be changed via the themeconfig.json file. When creating customized themes, you must therefore ensure these images include sufficient empty space starting from 755 pixels from the top of the image, in a contrasting color to the text, to allow the text to be visible.

* Added in v14.

Configuring colors, background and text

The wording and color of certain text and image backgrounds can be customized via the optional themeconfig.json file.

All colors in this file are specified using RGB hexadecimal notation (in the format 0xnnnnnn), apart from the following:

- indicator_num_bg_color

This color uses ARGB hexadecimal notation (0xnnnnnnnn), where the first two (optional) digits after the 0x represent the Alpha channel, indicating the level of opacity applied. Example values include:

- 00 - fully transparent
- 80 - 50% opaque
- 99 - 60% opaque
- BE - 75% opaque
- E6 - 90% opaque
- FF - fully opaque

themeconfig.json

The themeconfig.json contains the following. If any values **within** this file are not specified, Pexip Infinity will simply use the values from the Base theme.

```
{
    "enable_dtmf_conference_control" : true,
    "dtmf_conference_control_commands" : {
        "*7" : "toggle_lock",
        "*5" : "toggle_guest_mute",
        "##" : "end_conference"
    },

    "no_main_video_bg" : "0x000000",
    "no_presentation_video_bg" : "0x000000",

    "disable_streaming_indicator" : false,
    "disable_watermark_icon" : false,
    "streaming_indicator_text" : "Streaming enabled",

    "virtual_reception_text_color" : "0xFFFFFF",
```

```
    "plus_n_indicator_text_color" : "0xF38B3C",

    "indicator_num_color" : "0xF38B3C",
    "indicator_num_bg_color" : "0xBE222222"

    "test_call_service_media_delay" : 2,
    "test_call_service_disconnect_timeout" : 10
}
```

Item	Description	Value in Base theme
enable_dtmf_conference_control	Determines whether Host participants can use DTMF to control the conference (**true**) or not (**false**).	`true`

dtmf_conference_control_commands	If DTMF controls are enabled, this section specifies the DTMF digits used for each control. Current controls are:
	• conference lock and unlock (**toggle_lock**)
	• mute and unmute all guests (**toggle_guest_mute**)
	• terminate the conference (**end_conference**)
	When changing the DTMF controls, you must only edit the digits inside the first pair of quotes on each line (i.e. ***7**, ***5** and **##**). Editing anything else will disable DTMF commands.
	Each control must be two DTMF digits long, and all the commands must be different. See Changing DTMF controls for guidance on best practice.
		`"*7" : "toggle_lock"`
		`"*5" : "toggle_guest_mute"`
		`"##" : "end_conference"`

no_main_video_bg	The color of the background to the **no_main_video_icon.png** image.	`0x000000` (black)
no_presentation_video_bg	The color of the background to the **no_presentation_video_icon.png** image.	`0x000000` (black)
disable_streaming_indicator	Determines whether the streaming indicator is disabled (**true**) or enabled (**false**).	`false`
disable_watermark_icon	Determines whether the watermark icon is overlaid onto the main speaker video (**false**) or is not used (**true**). See Video watermarking for more information.	`false`
streaming_indicator_text	The text that appears to the right of the streaming indicator when it slides out over the main video image.	`Streaming enabled`
virtual_reception_text_color	The color of the text that is overlaid at the bottom of the **transfer.jpg** and **two_** images.	`0xFFFFFF` (white)
plus_n_indicator_text_color	The color of the text that is overlaid on the **plus_n_indicator.jpg** image, which shows how many additional participants are in the conference.	`0xF38B3C` (orange)
indicator_num_color	The color of the text that is overlaid on the **indicator_num_bg_color** background, which slides out to show how many additional audio participants are in the conference.	`0xF38B3C` (orange)
indicator_num_bg_color †	The color of the background to the **indicator_num_color** text, which slides out to show how many additional audio participants are in the conference.	`0xBE222222` (75% opaque dark gray)
test_call_service_media_delay *	The number of seconds that media is delayed before being looped back to the caller when using a Test Call Service. (For more information, see Configuring the Test Call Service.)	`2`

Item	Description	Value in Base theme
test_call_service_ disconnect_timeout *	The number of seconds that a user can test their media before the disconnect message is played, when using a Test Call Service.	10

* Added in v14.

† This value can optionally include a level of transparency using ARGB notation, where the first two digits represent the level of opacity applied (see Configuring colors, background and text for more information).

Base theme

Pexip Infinity ships with its own Base theme. Files from the Base theme are used when no other theme has been selected, or the selected theme does not contain the required file.

You cannot change the contents of the Base theme, but you can create your own themes by uploading customized versions of one or more of the Base files.

When a user accesses a service that has a customized theme applied, Pexip Infinity presents them with the sounds and images from the files included in that theme. If the customized theme does not include a specific file, or the service does not have a theme applied, then Pexip Infinity uses the relevant file from the default theme. If the default theme does not include the specific file either, then the file from the Base theme is used.

The same rules apply when deciding which Base theme file will be used. However, if any values **within** this file are not specified, Pexip Infinity will use the values from the Base theme's **themeconfig.json** file.

The files used in the Base theme are as described in File requirements for themes

Preconfigured themes

Pexip Infinity ships with a number of preconfigured themes. These themes can be copied and edited, making it easier for you to select and create themes suitable for your deployment.

The preconfigured themes are identical to the Base theme with the following exceptions:

- some include entry and exit tones (the Base theme contains "empty" tones files)
- some refer to the "#" key as "**the hash key**" (the Base theme refers to this as "**the pound key**").

The files that may differ from those in the Base theme are:

- **Entry tone:** conf-participant_entry_tone_48kHz_mono.wav
- **Exit tone:** conf-participant_exit_tone_48kHz_mono.wav
- **"#" key reference:** 2sd-number-pound-key_48kHz_mono.wav, conf-getpin_pound-key_48kHz_mono.wav, conf-waithostpin_ pound-key_48kHz_mono.wav

The content of the above files for each of the preconfigured themes is as follows:

Theme	Entry tone	Exit tone	"#" key references
Pexip theme (English_US) *	<empty file>	<empty file>	"...the pound key"
Pexip theme (English_UK)	<empty file>	<empty file>	"...the hash key"
Pexip theme (English_US) with entry tones	A high tone followed by a low tone	A low tone followed by a high tone	"...the pound key"
Pexip theme (English_UK) with entry tones	A high tone followed by a low tone	A low tone followed by a high tone	"...the hash key"

* This theme is identical to the Base theme, although unlike the Base theme it can be edited.

Pexip Infinity maintenance tasks

Typical maintenance tasks include upgrading the Pexip Infinity software, creating and restoring backups of configuration data, and managing administrator account passwords.

Setting and changing usernames and passwords

There are a number of different components of the Pexip Infinity platform that require a username and password:

- Pexip Infinity Administrator interface
- Management Node operating system
- Conferencing Node operating system

Information on how to set and change these is given below.

Some external systems such as CUCM also use usernames and passwords to authenticate with Pexip Infinity. For more information on these, see Integrating with external systems.

Pexip Infinity Administrator interface

A username and password are required for logging in to the Pexip Infinity Administrator interface. By default these credentials are verified against the local database and are the **web administration username** and **web administration password** that were set during initial installation (via the installation wizard). You can also configure the system to authenticate users against an LDAP-accessible database.

Local database authentication

The username (typically **admin**) cannot be changed after being set, but the password can.

To change the password used to log in to the Pexip Infinity Administrator interface:

1. Log in to the Pexip Infinity Administrator interface.
2. At the top right of the screen, select **Change password**.
3. Enter the existing password, and then enter the new password twice.
4. Select **Change my password**.

You will see a message:

Password change successful

Your password was changed.

Note that the **Change password** option is not available if the system has its **Authentication source** configured as ***LDAP database***.

LDAP-accessible database authentication

Usernames and passwords for the LDAP-accessible database cannot be managed via the Pexip Infinity Administrator interface.

Changing the Management Node operating system password

The Management Node virtual machine (VM) runs on a Linux operating system. The username is **admin** and cannot be changed.

The password for the Management Node operating system was set during initial installation of the Management Node (when running the installation wizard).

The password can be changed by logging in to the Management Node operating system over SSH (providing SSH access has not been disabled) and running the standard UNIX command `passwd`.

Conferencing Node operating system password

Conferencing Nodes run on a Linux operating system. The username for each Conferencing Node operating system is **admin** and cannot be changed.

The password was set during Deploying new Conferencing Nodes when configuring the **SSH password** field.

The password can be changed by logging in to the Conferencing Node operating system over SSH (providing SSH access has not been disabled) and running the standard UNIX command `passwd`.

Lost or forgotten passwords

If you forget the password for the Pexip Infinity Administrator interface, you can re-run the installation wizard, being sure to change only the **Web administration password** setting.

If you re-run the installation wizard to only reset the web administration password then you will not lose your configuration data and you will not need to delete and redeploy your Conferencing Nodes, providing you re-enter exactly the same hostname/addressing information as before. However, after completing the wizard you must then edit every system location (**Platform Configuration > Locations**) and reselect the DNS and NTP servers for that location.

If you forget the password used to log in to a Conferencing Node or Management Node operating system, contact your authorized Pexip representative for assistance.

Upgrading the Pexip Infinity platform

The upgrade process

When you initiate an upgrade of the Pexip Infinity software, the following steps occur automatically, in order.

1. The Management Node software is upgraded, after which the Management Node will automatically reboot.
2. The first Conferencing Node is put into maintenance mode, meaning all further incoming calls to it will be rejected.
3. When all calls have cleared from the first Conferencing Node, its software is upgraded and the system is rebooted. This process should take around 2 minutes to complete, but may take longer depending on the speed of the connection between the Management Node and Conferencing Node over which the files are transferred.
   If all calls have not cleared after 1 hour, the first Conferencing Node will be taken out of maintenance mode and put at the back of the queue of systems to be upgraded. A further attempt to upgrade the first Conferencing Node will be made after all other systems have been upgraded (or had upgrade attempts made).
4. After the first Conferencing Node has been upgraded successfully (or has been put back in the queue for a later upgrade attempt) and is again available, the second Conferencing Node is put into maintenance mode.
5. The process continues with each subsequent Conferencing Node being put into maintenance mode, and, after all calls have cleared, being upgraded and then rebooted.
   Any Conferencing Nodes used for dynamic cloud bursting will be automatically started up and upgraded.
6. If the upgrade of the Management Node and all Conferencing Nodes has not completed successfully after 24 hours, the process will stop and all systems will be left in their existing upgrade state. This is designed to prevent situations where one Conferencing Node cannot be upgraded, which would otherwise leave the system in a permanent state of upgrading.

If the upgrade process does not complete successfully and stops after 24 hours, you may have a mix of upgraded and non-upgraded systems. You will then need to repeat the upgrade process. During a repeat upgrade, only those systems that have not already been upgraded will be included in the upgrade process.

During the 24-hour period from when an upgrade has been initiated, you cannot re-initiate an upgrade using the Administrator interface. If you must re-initiate an upgrade during this time, you must reboot the Management Node and then start the process again.

When to upgrade

While the upgrade is in progress, some Conferencing Nodes will be running the newer version of the software and some will be running the older version. These Conferencing Nodes will be incompatible until they are all again running the same version. This means that there may be instances where two endpoints dial the same Virtual Meeting Room alias but if they are routed to

different Conferencing Nodes that are running different versions of the software, the two endpoints will be in different conferences. For this reason, we recommend upgrading at a time of minimal usage.

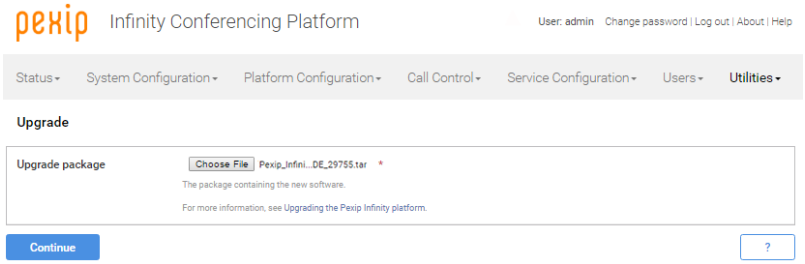
Upgrading from version 8 or later to version 14.1

If you need to upgrade your platform during active business hours we recommend that you put all of your Conferencing Nodes into maintenance mode prior to starting the upgrade. As each node is upgraded it will automatically come out of maintenance mode and be able to host new conferences.

During an upgrade from version 11 or earlier to version 12 or later you will not be able to host any conferences on those Conferencing Nodes that are still running the earlier software version. You will experience "No valid license available" errors should anybody try to join a conference on those nodes. Those nodes will be able to host conferences again as soon as they have been upgraded to the new software.

To upgrade Pexip Infinity software from v8 or later to v14.1:

1. Before upgrading an on-prem deployment, we recommend that you use your hypervisor's snapshot functionality to take a full VMware/Hyper-V snapshot of the Management Node. If upgrading from v8, due to incompatibilities resolved in v8.1, ensure that you take a non-quiescing snapshot. This snapshot will be required should you later need to downgrade, or if the upgrade fails. You may also want to take a snapshot of each Conferencing Node, although depending on the size and complexity of your deployment it may be easier to simply redeploy these from the Management Node in the unlikely event that this is required.
   Before upgrading a cloud-based deployment (AWS or Azure), you should backup the Management Node via Pexip Infinity's inbuilt mechanism (**Utilities > Backup/Restore**).
2. Download the Pexip Infinity upgrade package for v14.1 from **www.pexip.com/software-download**.
3. From the Pexip Infinity Administrator interface, go to **Utilities > Upgrade**.
4. Select **Choose File** and browse to the location of the upgrade package.



5. Select **Continue**. There will be a short delay while the upgrade package is uploaded.
   After the upgrade package has been uploaded, you are presented with a confirmation page showing details of the existing software version and the upgrade version.
6. To proceed, select **Start upgrade**.
   You are taken to the **Upgrade Status** page, showing the current upgrade status of the Management Node and all Conferencing Nodes (for a definition of each status, see Definition of upgrade statuses). This page automatically refreshes every 5 seconds.
7. When the upgrade completes, all systems will show a status of **No upgrade in progress** and have the new **Installed version**.
   If a Conferencing Node fails to upgrade, for example if it remains on a **Waiting for calls to clear** status, it should be rebooted. The upgrade process will then continue as expected.

No additional actions are required to upgrade the Management Node or Conferencing Nodes individually.

We recommend that you take a fresh backup of your system after upgrading.

Error messages during upgrade

The following Error-level messages are expected during an upgrade and do not require any action:

```
2014-03-03T12:12:01.924+00:00 <manager_hostname> 2014-03-03 12:12:01,924 Level="ERROR"
Name="administrator.system.configuration" Message="Node configuration failed."
Node="<worker_hostname>" Resource="File[/etc/certs/privatekey.pem]"
```

```
2014-03-03T12:12:01.924+00:00 <manager_hostname> 2014-03-03 12:12:01,924 Level="ERROR"
Name="administrator.system.configuration" Message="Node configuration failed."
Node=""<worker_hostname>" Resource="File[/etc/certs/server.pem]"

2014-03-03T12:12:02.400+00:00 <manager_hostname> 2014-03-03 12:12:02,399 Level="ERROR"
Name="administrator.system.connectivity" Message="Unable to contact node." Node="<node_
ip_fqdn>"
```

Upgrading from versions 3-7 to version 14.1

If you are running a Pexip Infinity software version between v3 and v7 inclusive, to upgrade to the latest version, you must first upgrade to version 8. To do this:

1. Download the Pexip Infinity v8 upgrade file (contact your Pexip authorized support representative for the link to this file).
2. Follow the steps outlined in Upgrading from version 8 or later to version 14.1, but when asked to **Choose File** browse to the location of the **v8** upgrade file.
3. Verify that the upgrade has completed successfully.
4. Download the Pexip Infinity **v14.1** upgrade file.
5. Follow the steps outlined in Upgrading from version 8 or later to version 14.1, and when asked to **Choose File** browse to the location of the **v14.1** upgrade file.

Upgrading configuration-only deployments

The automatic upgrade process described above will update the Management Node and all Conferencing Nodes, including Conferencing Nodes that have been created using the configuration only deployment type.

However, if after the upgrade you wish to deploy new configuration-only Conferencing Nodes, you must download and use a version of the Conferencing Node VM template that matches the version of the Management Node that you have upgraded to. Creating a new Conferencing Node from a VM template containing a different version of the Pexip Infinity software than that which is running on the Management Node is not supported and will not work. For more information, see Deploying a Conferencing Node using a generic VM template and configuration file

Definition of upgrade statuses

During an upgrade, the **Upgrade Status** page will report the status of each Conferencing Node as follows:

Status	Description
No upgrade in progress	During a platform upgrade, this is the default state that occurs before a Conferencing Node is upgraded, or after the node has rebooted .
Upgrade pending	An upgrade of the platform is in progress and this Conferencing Node is in the queue to be upgraded.
Preparing to upgrade	The Conferencing Node is preparing to upgrade. During this time the node will be put into maintenance mode, and other services will be stopped.
Waiting for calls to clear	The Conferencing Node is in maintenance mode and is waiting for existing conferences to complete.
Timeout waiting for calls to clear	Not all conferences had cleared after 1 hour. This Conferencing Node has been removed from maintenance mode and the upgrade will be attempted again later.
Upgrade in progress	The new software is being unpackaged and installed on the Conferencing Node. During this time the Conferencing Node will not synchronize configuration with the Management Node.
Rebooting	The upgrade has completed and the Conferencing Node is rebooting.
Could not communicate with conferencing node	This error will be reported if the Conferencing Node cannot be contacted.

Downgrading or recovering from a failed upgrade

To downgrade your system to the previous version, or to recover your system after a failed upgrade:

1. Restore the Management Node from the VMware snapshot that you took using your hypervisor at the start of the upgrade process.
2. Restore the individual Conferencing Nodes by either:
   - redeploying each Conferencing Node manually from the Management Node - see Deploying new Conferencing Nodes, or
   - restoring the hypervisor snapshot of each Conferencing Node that was taken at the start of the upgrade process.

Checking the installed version

Management Node

To view which software version is running on the Management Node, click on the **About** link at the top right corner of the Pexip Infinity Administrator interface.

Conferencing Nodes

To view which software version is running on individual Conferencing Nodes, go to **Status > Conferencing Nodes**. The software version number will be shown in the **Installed version** column.

Re-running the installation wizard

You can rerun the installation wizard if you previously exited the wizard before completing all the steps, or you entered the wrong hostname/addressing information (for server-based deployments) during the initial configuration.

Re-running the installation wizard will overwrite any existing Management Node configuration and lose all connections to existing Conferencing Nodes. Therefore, if you re-run the installation wizard after deploying any Conferencing Nodes, you must delete any existing Conferencing Nodes and then redeploy them from the updated Management Node.

If you want to change the IP address of the Management Node you must first return your licenses from your current Management Node. If you do not return your licenses before re-running the installation wizard you will lose your licenses from your new Management Node (with the new IP address) and you will not be able to add them back to that new Management Node. See Moving, restoring or changing the IP address of the Management Node for instructions on how to do this.

Do not re-run the installation wizard on cloud-based deployments (AWS or Azure) in order to change Management Node configuration data such as its IP address or hostname. To change such data you must terminate the existing instance and deploy a new Management Node instance. You should only re-run the installation wizard on cloud-based deployments if you need to reset the web administration password (and then you should not change any of the other configuration data).

If you re-run the installation wizard to only reset the web administration password then you will not lose your configuration data and you will not need to delete and redeploy your Conferencing Nodes, providing you re-enter exactly the same hostname/addressing information as before. However, after completing the wizard you must then edit every system location (**Platform Configuration > Locations**) and reselect the DNS and NTP servers for that location.

To re-run the installation wizard:

1. Open a console window on the Management Node VM.
   If your Management Node is deployed on AWS or Azure, use an SSH client to access the Management Node, supplying your private key file if appropriate.
2. At the login prompt, log in as **admin**.
3. If requested, enter the operating system password created when the installation wizard was run. If you did not get as far as creating a new password, you will be asked to create one now.
4. Type `installwizard`.
   If your Management Node is deployed on AWS or Azure, you will be prompted again for the admin password.
   The Pexip installation wizard will begin.

(Note that the steps described above can vary slightly depending on whether it is a cloud or server-based deployment, and whether you previously completed all of the installation wizard steps.)

Follow the prompts to set the following configuration for the Management Node.

If you press enter, the default value will be applied:

Setting	Default value	Multiple entries allowed?	Can be changed via Pexip Infinity Administrator interface?
IP address	192.168.0.100 *	No	No ‡
Network mask	255.255.255.0 *	No	No ‡
Gateway	192.168.0.1 *	No	No ‡
Hostname	<no default>	No	No ‡
Domain suffix	<no default>	No	No ‡
DNS servers	8.8.8.8	Yes, if separated by a space	Yes
NTP servers †	Two of the following: • 0.pexip.pool.ntp.org • 1.pexip.pool.ntp.org • 2.pexip.pool.ntp.org • 3.pexip.pool.ntp.org	Yes, if separated by a space	Yes
Web administration username	admin	No	No ‡
Web administration password	<no default>	No	Yes
Enable incident reporting (yes/no)	<no default>		Yes
Send deployment and usage statistics to Pexip (yes/no)	<no default>		Yes

* These defaults apply only if your network does not have DHCP enabled. If DHCP is enabled, the defaults will be those allocated by the DHCP server. The addresses you enter using this wizard will be assigned as static IP addresses.

* When deploying on AWS or Microsoft Azure these values will be replaced with the IP address and network settings for your instance.

† The NTP server must be accessible by the Management Node at the time the startup wizard is run. Installation will fail if the Management Node is unable to synchronize its time with an NTP server.

‡ After they have been configured, do not attempt to change these settings by any other means. To change these settings on server-based deployments, you must re-run the installation wizard. To change these settings on cloud-based deployments (AWS or Azure), you must terminate the existing instance and deploy a new Management Node instance.

Migrating Conferencing Nodes between host VMware servers

Using the tools available in VMware, you can move Conferencing Nodes from one host server to another.

You may want to do this if the existing host server is running low on capacity or is experiencing performance issues.

Pexip Infinity supports VMware's vMotion for the live migration of Conferencing Nodes.

You must put the Conferencing Node into maintenance mode and wait until all conferences on that node have finished before migrating it to another host server. See Taking a Conferencing Node out of service for more information.

Prerequisites

The source and target servers must be enabled for vMotion and use the same shared storage.

Manual migration via VMware's vMotion

1. Ensure that the Conferencing Node is in maintenance mode and that all conferences on that node have finished.
2. In the VMware inventory view, right-click on the Conferencing Node and select **Migrate**. The **Migrate Virtual Machine** window will appear.
3. Select **Change host** then select **Next**.
4. Select the destination host server or cluster and then select **Next**.
5. Select the destination resource pool and then select **Next**.
6. Select the priority and then select **Next**.
7. Check the details of the migration, and if they are all correct, select **Finish**.

Taking a Conferencing Node out of service

If you need to take a Conferencing Node out of service, you must first put it into maintenance mode to ensure that it is not hosting any conferences.

To do this:

1. Go to **Platform Configuration > Conferencing Nodes**.
2. Select the Conferencing Node.
3. Select the **Enable maintenance mode** check box and select **Save**.
   While maintenance mode is enabled, this Conferencing Node will not accept any new conference instances.
4. Wait until any existing conferences on that Conferencing Node have finished. To check, go to **Status > Live View**.

When all conferences on the Conferencing Node have finished you can take it out of service.

Note that when a Conferencing Node is in maintenance mode, it reports a media load of 100%. This is to indicate that there is no current capacity available.

About maintenance mode

When a Conferencing Node goes into maintenance mode, it will not accept any new calls. Any existing calls on that Conferencing Node will not be affected. After all existing calls have terminated, the Conferencing Node will still be live and contactable but will not be handling any calls.

A Conferencing Node will go into maintenance mode for one of three reasons:

1. The system administrator has elected to put the Conferencing Node into maintenance mode. This is done from the Pexip Infinity Administrator interface by going to **Platform Configuration > Conferencing Nodes**, selecting the relevant Conferencing Node, and selecting the **Enable maintenance mode** checkbox.
   This setting will persist after a reboot.
2. The Pexip Infinity platform is being upgraded, during which time each Conferencing Node in turn is automatically put into maintenance mode and upgraded. For more information, see Upgrading the Pexip Infinity platform.
3. The Conferencing Node has been installed on a system that does not meet the CPU instruction set requirements. In such a case:
   - after initial deployment, the following message will appear in the admin log:

     ```
     CPU instruction set is not supported; system will be placed in maintenance mode
     ```

   - each time the Conferencing Node rejects a call, the following message will appear in the admin log:

     ```
     Message="Participant failed to join conference." Reason="System in maintenance mode"
     ```

   - any manual changes to the **Enable maintenance mode** setting will have no impact - the Conferencing Node will remain in maintenance mode regardless of this setting.

   For more information on how to resolve this, see Troubleshooting the Pexip Infinity platform.

Rebooting a Conferencing Node

Occasionally, it may be necessary to reboot a Conferencing Node. A reboot is typically required if:

- You have changed some specific **Global Settings** such as the media and signaling port ranges, or you have enabled or disabled call protocols.
- A Conferencing Node fails to upgrade, for example if it remains on a **Waiting for calls to clear** status.

There are two main ways to reboot a Conferencing Node:

- Log in to the hypervisor or your cloud service that is managing the host server and reboot the virtual machine using the relevant processes according to the management system.
- Connect to the Conferencing Node via SSH, log in as admin, and issue the command `sudo reboot`.

Backing up and restoring configuration

You should take regular backups of the configuration data on your Management Node, particularly before and after Upgrading the Pexip Infinity platform. This allows you to restore your configuration to a specific point in time, or to restore your configuration if you have to deploy a new Management Node.

There are 2 ways to maintain copies of your Management Node configuration data:

- use your hypervisor's tools to create a full snapshot of the Management Node VM
- use the backup and restore mechanism built into the Pexip Infinity Administrator interface (automatic and manual backup options are available)

For on-prem deployments, we recommend that you use both methods to preserve your configuration data. The VM snapshot should be your primary mechanism, as this allows you to easily restore your configuration back to its state at the time the snapshot was taken. The Pexip Infinity backup and restore mechanism is your fallback mechanism, as this allows you to preserve a copy of your data in an alternative location, in case you lose your VM environment. Cloud-based deployments (AWS and Azure) should use the Pexip Infinity backup and restore mechanism.

You can also separately backup and restore just your Virtual Meeting Room and Virtual Auditorium configuration, see Bulk import/export of service configuration.

Backing up Conferencing Nodes

Conferencing Nodes do not need to be backed up. They receive all their configuration information from the Management Node and can simply be redeployed if necessary. However, if your Conferencing Nodes are geographically spread out and redeploying them would consume significant bandwidth or take a significant length of time, they can also be backed up with your hypervisor's backup tools.

Using your hypervisor's tools to take and restore snapshots

You can use your hypervisor's own backup tools, or any other third-party tool that supports VM backups, to create a full snapshot of the Management Node. This snapshot can then be re-deployed at a later date if required.

Note that snapshots can take a significant amount of storage space. Therefore, we recommend taking snapshots only when necessary (such as prior to an upgrade) and deleting the snapshot as soon as possible.

You should only create and delete VMware snapshots at a time of minimal usage. Taking or removing snapshots can cause virtual machines to become unresponsive.

Consult your hypervisor documentation for more information about taking and restoring snapshots.

Using backup and restore via the Pexip Infinity Administrator interface

You can use Pexip Infinity's inbuilt backup and restore mechanism to backup and restore the configuration data on the Management Node.

You can enable daily automatic backups, and you can also take a manual backup whenever it is appropriate, for example, before and after you make any configuration changes or perform a software upgrade.

- All backup files are encrypted — the administrator supplies a passphrase and must remember this for any subsequent restoration.
- Restoration must occur on exactly the same version that the backup was taken from.
- The data contained in the backup contains all configuration data, including IP addresses, custom themes, certificates and call history.
- The backup data does not contain licenses, the administrator log, the support log, usage statistics or the operating system password.

Note that this function can only be used to replicate the configuration of a previous Management Node onto a new node. It cannot be used to redeploy Conferencing Nodes.

Enabling daily automatic backups

You can enable Pexip Infinity to automatically backup the Management Node configuration data on a daily basis.

When automatic backups are enabled:

- Each backup is taken at 01:02 UTC every day.
- Successful backup operations are recorded in the administrator log (with a "Created automatic system backup" message).
- Automatic backup filenames take the format:
  pexip_auto_backup_<hostname>_<version>_<build>_<date>_<time>.tar.gpg
- You can see the set of backup files that are currently stored on the host VM by going to **Utilities > Backup/Restore** and looking at the list of **Existing Backup Files**.
- If you want to download a backup file from the host VM to another machine, you have to do this manually (by selecting the **Download backup** option from the **Backup/Restore** page).
- The system keeps on the host VM only the 5 most recent manually-taken backups and the 5 most recent automatic backups. Older backup files are deleted.

To enable automatic backups:

1. Go to **Platform Configuration > Global Settings**.
2. Select the **Enable automatic backups** checkbox.
3. Enter a **Backup passphrase**.
   The text entered here is used to encrypt the backup file. You must remember this text as it will be required if you need to subsequently restore the data from the file.
4. Select **Save**.

Manually creating a backup file

To manually create and download a backup file:

1. Go to **Utilities > Backup/Restore**.
2. In the **Create backup** section, enter a **Passphrase** and then enter it again in the **Re-enter passphrase** field.
   The text entered here is used to encrypt the backup file. You must remember this text as it will be required if you need to subsequently restore the data from the file.
3. Select **Create backup**.
   After a few seconds you will see a message: "Successfully created the backup file: <file_name>" where <file_name> takes the format:
   pexip_backup_<hostname>_<version>_<build>_<date>_<time>.tar.gpg
4. Download the file from the host VM:
   a. From the **Existing backup files** section at the bottom of the page, select **Download backup** for the file you have just created.
   b. Follow your browser's prompts to save or download the file to your local file system.
   c. If required, you can delete unwanted backup files from your host VM by selecting **Delete backup**.
      The system keeps on the host VM only the 5 most recent manually-taken backups and the 5 most recent automatic backups. Older backup files are deleted.

Restoring data to the Management Node

You can restore configuration data to the Management Node. This could be restored to the original Management Node from which the backup was taken, or it could be restored to a newly deployed Management Node (if the original node was lost due to, for example, issues with its VM environment).

Any in-progress calls will not be affected while data is being restored to the Management Node.

To restore configuration data:

1. If required, deploy a new Management Node (see Installation overview for links to the appropriate instructions for your hypervisor).
   - Complete the installation wizard as normal.
   - The IP address you enter at this stage is temporary (it can be the same as the previous Management Node).
   - All the configuration data you enter, including the IP address, will be subsequently replaced.
2. If any new Conferencing Nodes have been deployed since the backup was taken:
   - Power these Conferencing Nodes off and delete them before restoring the Management Node data.
   - These additional Conferencing Nodes will not be recognized by the restored Management Node. You will have to create them again after the restore has completed.
3. If previously configured Conferencing Nodes have been deleted since the backup was taken:
   - There is no need to do anything at this stage; the restore will complete successfully.
   - Simply delete the Conferencing Nodes from the restored configuration, after the restore has completed.
4. Restore your previous backup file:
   a. Go to **Utilities > Backup/Restore**.
   b. In the **Restore backup** section, enter the **Passphrase**.
      The text entered here must be identical to the text that was used to create the backup file.
   c. Select **Choose File** and then choose the backup file that you want to restore.
      The file must be chosen from your local file system (you cannot select a file from the list of **Existing backup files**).
   d. Select **Restore backup**.
      You are taken to the **Restore Backup** confirmation page.
      If instead, you see "Failed to restore the system from the backup file" with a "Decryption Error: decryption failed" message, the most likely reason for this is that you have entered an incorrect passphrase.
   e. The confirmation page shows the date that the backup was taken, and the Management Node IP address that will be restored to this system.
      Select **Restore backup** to confirm the restoration.
5. If the restore is successful, after a few seconds you will see a "Successfully restored the backup file" message and the Management Node will reboot.
   You will need to wait for the node to return from rebooting before you can access it again.
   If you have restored a file where the IP address of the Management Node in the backup file is different from the node's current address (prior to the restore), you will need to manually enter this IP address into the web browser to access the restored Management Node's web interface login page.
6. If you have restored your configuration onto a new VM (and were unable to return the licenses from your previous VM), you must contact your Pexip authorized support representative to get your licenses reapplied on your new VM.

Moving, restoring or changing the IP address of the Management Node

These procedures explain how to change the IP address of your Management Node, or physically move your Management Node to a new host server, without losing your Pexip Infinity platform configuration or invalidating your Pexip Infinity licenses.

Before performing any of these procedures we recommend that you follow standard IT practices, as per your company policy. For example, ensure that you take a snapshot of the Management Node prior to making any changes, and if possible take offline backups of the VMs.

In all cases you will have a loss of service while you perform the maintenance procedure.

Note that these instructions only apply to on-premises deployments. For cloud-based deployments, if you need to change the IP address of the Management Node you must decommission your entire deployment, terminating all of the existing instances, and then redeploy your system.

Changing the IP address of the Management Node (same physical host)

To change the IP address of your Management Node, where you want to keep the same physical host for the Management Node:

1. We recommend that you backup your current configuration:
   a. Go to **Utilities > Backup/Restore**.
   b. In the **Create backup** section, enter a **Passphrase** and then enter it again in the **Re-enter passphrase** field.
      The text entered here is used to encrypt the backup file. You must remember this text as it will be required if you need to subsequently restore the data from the file.
   c. Select **Create backup**.
      After a few seconds you will see a message: "Successfully created the backup file: <file_name>" where <file_name> takes the format:
      pexip_backup_<hostname>_<version>_<build>_<date>_<time>.tar.gpg
   d. Download the file from the host VM:
      i. From the **Existing backup files** section at the bottom of the page, select **Download backup** for the file you have just created.
      ii. Follow your browser's prompts to save or download the file to your local file system.
      iii. If required, you can delete unwanted backup files from your host VM by selecting **Delete backup**.
      The system keeps on the host VM only the 5 most recent manually-taken backups and the 5 most recent automatic backups. Older backup files are deleted.
2. Return your Pexip Infinity license.
   You must return your license before decommissioning your current host installation.
   a. Go to **Platform Configuration > Licenses**.
   b. Select the license you want to deactivate.
   c. Make a note of the **License entitlement key**.
   d. Select **Return license**.
3. Decommission your Conferencing Nodes i.e. delete the Conferencing Nodes and remove the associated VMs. Follow this procedure for each Conferencing Node:
   a. Put the Conferencing Node into maintenance mode and wait until all calls on it have terminated.
   b. From the Pexip Infinity Administrator interface, go to **Platform Configuration > Conferencing Nodes**.
   c. Select the Conferencing Node to be deleted, and from the **Action** drop-down menu select ***Delete selected Conferencing Nodes***.
   d. Select **Go** and on the following page confirm that you want to delete the selected Conferencing Node by selecting **Yes, I'm sure**.
   e. Shut down and remove the Conferencing Node VM.
      i. Log in to the VM manager, shut down the deleted Conferencing Node and then power it off.
      ii. Right-click on the Conferencing Node and select ***Delete from Disk*** (VMware) or ***Delete*** (Hyper-V / KVM / Xen).
4. Re-run the installation wizard and supply the new IP address of the Management Node:
   a. Open a console window on the Management Node VM.
   b. At the login prompt, log in as **admin**.
   c. Enter the operating system password.
   d. Type `installwizard`.
      The Pexip installation wizard will begin.
      At the **IP address** prompt, enter your new Management Node IP address.

5. Reapply your previous license:
   a. Go to **Platform Configuration > Licenses**.
   b. Select **Add License**.
   c. In the **License entitlement key** field, enter the entitlement key from the 'old' Management Node that you noted previously.
   d. Select **Save**.

   The system will attempt to activate the license on the 'new' Management Node:
   - If the license is activated successfully, you will be returned to the **Licensing** page and will see the new license under the **Licensing** section.
   - If the activation attempt is unsuccessful (for example, if the Management Node was unable to establish a connection to the Pexip licensing server), or you selected **Manually activate**, the license will be saved as a **Stored license request**. You must then activate it manually.
6. Re-provision your Conferencing Nodes.

   See Deploying new Conferencing Nodes for links to the appropriate instructions for your hypervisor.

Moving/restoring a Management Node to a new physical host (same IP address)

Note that if your Management Node is installed on VMware and you do not need to perform a fundamental redeployment of the Management Node, i.e. you only want to move it across host servers, then you should use vMotion to perform the move.

To move or restore a Management Node to a new physical host, where you are keeping the same IP address for the Management Node on the new host server:

1. Backup your current configuration:
   a. Go to **Utilities > Backup/Restore**.
   b. In the **Create backup** section, enter a **Passphrase** and then enter it again in the **Re-enter passphrase** field.

      The text entered here is used to encrypt the backup file. You must remember this text as it will be required if you need to subsequently restore the data from the file.
   c. Select **Create backup**.

      After a few seconds you will see a message: "Successfully created the backup file: <file_name>" where <file_name> takes the format:

      pexip_backup_<hostname>_<version>_<build>_<date>_<time>.tar.gpg
   d. Download the file from the host VM:
      i. From the **Existing backup files** section at the bottom of the page, select **Download backup** for the file you have just created.
      ii. Follow your browser's prompts to save or download the file to your local file system.
      iii. If required, you can delete unwanted backup files from your host VM by selecting **Delete backup**.

      The system keeps on the host VM only the 5 most recent manually-taken backups and the 5 most recent automatic backups. Older backup files are deleted.
2. Return your Pexip Infinity license.

   You must return your license before decommissioning your current host installation.
   a. Go to **Platform Configuration > Licenses**.
   b. Select the license you want to deactivate.
   c. Make a note of the **License entitlement key**.
   d. Select **Return license**.
3. Install the Management Node on the new host server.

   Note that all of the configuration data that you enter via the installation wizard, including the IP address, will be subsequently replaced when you restore your backup.

   See Installation overview for links to the appropriate instructions for your hypervisor.
4. Restore the previous configuration to the new Management Node from your backup.

a. Go to **Utilities > Backup/Restore**.
b. In the **Restore backup** section, enter the **Passphrase**.
   The text entered here must be identical to the text that was used to create the backup file.
c. Select **Choose File** and then choose the backup file that you want to restore.
   The file must be chosen from your local file system (you cannot select a file from the list of **Existing backup files**).
d. Select **Restore backup**.
   You are taken to the **Restore Backup** confirmation page.
   If instead, you see "Failed to restore the system from the backup file" with a "Decryption Error: decryption failed" message, the most likely reason for this is that you have entered an incorrect passphrase.
e. The confirmation page shows the date that the backup was taken, and the Management Node IP address that will be restored to this system.
   Select **Restore backup** to confirm the restoration.

If the restore is successful, after a few seconds you will see a "Successfully restored the backup file" message and the Management Node will reboot.
You will need to wait for the node to return from rebooting before you can access it again.

5. Reapply your previous license:
   a. Go to **Platform Configuration > Licenses**.
   b. Select **Add License**.
   c. In the **License entitlement key** field, enter the entitlement key from the 'old' Management Node that you noted previously.
   d. Select **Save**.

   The system will attempt to activate the license on the 'new' Management Node:
   - If the license is activated successfully, you will be returned to the **Licensing** page and will see the new license under the **Licensing** section.
   - If the activation attempt is unsuccessful (for example, if the Management Node was unable to establish a connection to the Pexip licensing server), or you selected **Manually activate**, the license will be saved as a **Stored license request**. You must then activate it manually.

Moving/restoring a Management Node to a new physical host (different IP address)

To manually move or restore a Management Node to a new physical host server, where you also want to use a different IP address for the Management Node:

1. Backup your current configuration:
   a. Go to **Utilities > Backup/Restore**.
   b. In the **Create backup** section, enter a **Passphrase** and then enter it again in the **Re-enter passphrase** field.
      The text entered here is used to encrypt the backup file. You must remember this text as it will be required if you need to subsequently restore the data from the file.
   c. Select **Create backup**.
      After a few seconds you will see a message: "Successfully created the backup file: <file_name>" where <file_name> takes the format:
      pexip_backup_<hostname>_<version>_<build>_<date>_<time>.tar.gpg
   d. Download the file from the host VM:
      i. From the **Existing backup files** section at the bottom of the page, select **Download backup** for the file you have just created.
      ii. Follow your browser's prompts to save or download the file to your local file system.
      iii. If required, you can delete unwanted backup files from your host VM by selecting **Delete backup**.
      The system keeps on the host VM only the 5 most recent manually-taken backups and the 5 most recent automatic backups. Older backup files are deleted.

2. Return your Pexip Infinity license.
   You must return your license before decommissioning your current host installation.
   a. Go to **Platform Configuration > Licenses**.
   b. Select the license you want to deactivate.
   c. Make a note of the **License entitlement key**.
   d. Select **Return license**.
3. Decommission your Conferencing Nodes i.e. delete the Conferencing Nodes and remove the associated VMs. Follow this procedure for each Conferencing Node:
   a. Put the Conferencing Node into maintenance mode and wait until all calls on it have terminated.
   b. From the Pexip Infinity Administrator interface, go to **Platform Configuration > Conferencing Nodes**.
   c. Select the Conferencing Node to be deleted, and from the **Action** drop-down menu select ***Delete selected Conferencing Nodes***.
   d. Select **Go** and on the following page confirm that you want to delete the selected Conferencing Node by selecting **Yes, I'm sure**.
   e. Shut down and remove the Conferencing Node VM.
      i. Log in to the VM manager, shut down the deleted Conferencing Node and then power it off.
      ii. Right-click on the Conferencing Node and select ***Delete from Disk*** (VMware) or ***Delete*** (Hyper-V / KVM / Xen).
4. Install the Management Node on the new host server.
   All the configuration data you enter via the installation wizard, including the IP address, will be subsequently replaced when you restore your backup. At this stage you can use the same IP address as you used for your previous Management Node installation.
   See Installation overview for links to the appropriate instructions for your hypervisor.
5. Restore the previous configuration to the new Management Node from your backup (this will also restore the original Management Node IP address — you will set the new address in the next step):
   a. Go to **Utilities > Backup/Restore**.
   b. In the **Restore backup** section, enter the **Passphrase**.
      The text entered here must be identical to the text that was used to create the backup file.
   c. Select **Choose File** and then choose the backup file that you want to restore.
      The file must be chosen from your local file system (you cannot select a file from the list of **Existing backup files**).
   d. Select **Restore backup**.
      You are taken to the **Restore Backup** confirmation page.
      If instead, you see "Failed to restore the system from the backup file" with a "Decryption Error: decryption failed" message, the most likely reason for this is that you have entered an incorrect passphrase.
   e. The confirmation page shows the date that the backup was taken, and the Management Node IP address that will be restored to this system.
      Select **Restore backup** to confirm the restoration.
   If the restore is successful, after a few seconds you will see a "Successfully restored the backup file" message and the Management Node will reboot.
   You will need to wait for the node to return from rebooting before you can access it again.
6. Re-run the installation wizard and supply the new IP address of the Management Node:
   a. Open a console window on the Management Node VM.
   b. At the login prompt, log in as **admin**.
   c. Enter the operating system password.
   d. Type `installwizard`.
      The Pexip installation wizard will begin.
      At the **IP address** prompt, enter your new Management Node IP address.

7. Reapply your previous license:
   a. Go to **Platform Configuration > Licenses**.
   b. Select **Add License**.
   c. In the **License entitlement key** field, enter the entitlement key from the 'old' Management Node that you noted previously.
   d. Select **Save**.

   The system will attempt to activate the license on the 'new' Management Node:
   - If the license is activated successfully, you will be returned to the **Licensing** page and will see the new license under the **Licensing** section.
   - If the activation attempt is unsuccessful (for example, if the Management Node was unable to establish a connection to the Pexip licensing server), or you selected **Manually activate**, the license will be saved as a **Stored license request**. You must then activate it manually.
8. Re-provision your Conferencing Nodes.

   See Deploying new Conferencing Nodes for links to the appropriate instructions for your hypervisor.

Downloading a diagnostic snapshot

If you are experiencing issues with your Pexip Infinity service, you may be asked by your Pexip authorized support representative to provide them with a snapshot of your system to assist them in diagnosis of the issue.

To download the diagnostic snapshot:

1. On the Pexip Infinity Administrator interface, go to **Utilities > Diagnostic Snapshot**.
2. To download all available data, select **Download full snapshot**.

   or

   To download up to 24 hours of data:
   a. Type in, or adjust the slider to the number of hours of diagnostic data to be downloaded.
   b. Select **Download limited duration snapshot**.
3. Follow your browser's prompts to save the file.

File contents

The diagnostic snapshot is a collection of logs and incident reports and contains information such as IP addresses, conference names, aliases, Pexip Infinity configuration and system logs. Some of this information may be sensitive to your company, so the snapshot should be saved and handled securely.

File size

When each log reaches its maximum size, the oldest data is overwritten.

The maximum file size of each log in the snapshot is as follows:

- Administrator: 0.5 GB
- Support: 0.5 GB
- Internal developer-specific logs: 2 GB

The diagnostic snapshot also includes all available incident reports:

- the full snapshot contains up to 28 days of incident reports, with a cap at 5 GB
- the limited duration snapshot contains incident reports from the selected number of hours.

Automatically reporting errors

In order to allow continual monitoring and improvement of the Pexip Infinity platform, we encourage customers to send us details of any incidents that occur in their products. When this feature is enabled, incident reports are sent automatically to a specified URL. By default this is the address of a secure web server owned and managed by Pexip, but you have the option to change the URL.

We collate and analyze this information in order to identify issues with the software that can be resolved in future product releases. The information we receive will not be used for any other purpose.

The feature is enabled and disabled on a Pexip Infinity platform-wide basis, but when enabled each Management Node and Conferencing Node will send their individual incident reports directly to the incident reporting server.

You must therefore ensure that your IP routing and firewall configuration allows outbound communications from each of these systems to the nominated URL. By default, reports are sent to **https://acr.pexip.com** over TLS to port 443.

Content of incident reports

The exact content of each incident report will vary depending on the nature of the problem, but all reports **always** include:

- the date and time that the incident occurred
- the FQDN of the system on which the incident occurred
- the IP address of the system on which the incident occurred
- information about the Pexip Infinity software version
- details of the internal software processes running at the time the incident occurred.

The incident reporting server will also annotate each report with the IP address from which it was received (generally this will be the public IP address of the sender's network). This information is not contained in the original incident report.

The incident reports **may** also include:

- system configuration information
- stack traces.

The incident reports **never** include:

- system logs.

Enabling and disabling automatic sending of incident reports

Automatic sending of incident reports is enabled or disabled during initial installation of the Management Node (when running the installation wizard).

To enable or disable the automatic submission of incident reports, or to change the URL to which they are sent:

1. On the Pexip Infinity Administrator interface, go to **Platform Configuration > Global Settings**.
2. Select or deselect the **Enable incident reporting** box as required.
3. If appropriate, in the **Incident reporting URL** field enter the URL to which reports will be sent.
4. Select **Save**.

Automatically sending usage statistics

As part of our desire to continuously improve operation of the Pexip Infinity platform based on real-life customer usage data, we have enabled the ability for customers, at their own discretion, to automatically provide us with statistics relating to the usage and configuration of their deployment.

When enabled, relevant conference information (such as conference duration and number of participants, call duration and protocol) and selected configuration settings (such as version, license information, location and number of conferencing nodes) is sent automatically every 30 minutes to a third party service provider on behalf of Pexip. If you wish, we are happy to provide you with regular reports containing an analysis of your information, giving you insight into the usage patterns within your organization and helping you in your long-term capacity and growth planning.

We will also anonymize, aggregate, and analyze the information from all customers who have enabled this option, in order to better understand the ways our products are being used in production deployments. This in turn helps us determine the features and requirements of future software releases, which will ultimately provide a product that is of maximum benefit to you.

The analytics information we receive will not be used for any other purpose, will not contain any identifying information (such as user names or passwords), and will not be shared with any other organizations in an identifiable manner. A full list of the data that

will be sent to us is given in Information sent when usage reporting is enabled. For more information, please contact your Pexip authorized support representative.

When this option is enabled, your firewall must be configured to allow the Management Node to send outbound traffic over HTTPS.

Enabling and disabling automatic sending of usage statistics

For new deployments, automatic sending of usage statistics is enabled or disabled during initial installation of the Management Node (when running the installation wizard).

For upgrades from existing deployments running Pexip Infinity v7 or earlier, this feature will be disabled by default.

To enable or disable the automatic submission of usage statistics after initial installation or upgrade:

1. On the Pexip Infinity Administrator interface, go to **Platform Configuration > Global Settings**.
2. Select or deselect the **Automatically send deployment and usage statistics to Pexip** box as required.
3. Select **Save**.

Infinity Connect usage statistics

Individual Infinity Connect users can also control via their Infinity Connect application's **Settings** page whether to send usage statistics.

Note that the **Automatically send deployment and usage statistics to Pexip** global setting on the Management Node must also be enabled in order to allow the Infinity Connect application to send usage statistics.

Bulk import/export of service configuration

If you want to configure your Pexip Infinity platform with a large number of Virtual Meeting Rooms, Virtual Auditoriums or Virtual Receptions, you can import the configuration for each of those services from a CSV file.

This feature allows you to import all configuration apart from the Automatically Dialed Participants (ADPs) to be associated with each service. These can be imported separately; see Bulk import/export of Automatically dialed participants (ADPs). You can also import the aliases that devices are allowed to register with; see Bulk import/export of device aliases.

You can import Virtual Meeting Rooms from directory information contained in a Windows Active Directory LDAP server, or any other LDAP-accessible database. For more information, see Provisioning VMRs and devices from Active Directory via LDAP.

You can also export all of your existing Virtual Meeting Room, Virtual Auditorium or Virtual Reception configuration data to a CSV file. You may want to do this for backup purposes or to transfer configuration between, for example, a test system and a production system.

Preparing the CSV file for import

Header row

A header row in the CSV file is optional. If included, it should use exactly the same text for each field name as shown in the following sections, although the order of the fields may vary.

Format

The data in the CSV file must use the following format, and fields must be in the same order as shown (unless you have used a header file):

Virtual Meeting Room

```
name,description,pin,allow_guests,guest_pin,alias_alias,alias_description,tag,max_callrate_in,max_
callrate_out,call_type,host_view,enable_overlay_text,ivr_theme_name,participant_limit,primary_owner_
email_address
```

Virtual Auditorium

```
name,description,pin,allow_guests,guest_pin,alias_alias,alias_description,tag,max_callrate_in,max_
callrate_out,call_type,host_view,guest_view,force_presenter_into_main,enable_overlay_text,ivr_theme_
name,participant_limit,mute_all_guests
```

Virtual Reception

```
name,description,alias_alias,alias_description,tag,max_callrate_in,max_callrate_out,call_type,ivr_theme_
name,mssip_proxy_name,match_string,replace_string,system_location_name
```

where:

name	The name used to refer to this Virtual Meeting Room, Virtual Auditorium or Virtual Reception. This field cannot be left blank.
description	An optional description of the Virtual Meeting Room, Virtual Auditorium or Virtual Reception.
pin	(Virtual Meeting Room and Virtual Auditorium only) This optional field allows you to set a secure access code that must be entered by participants before they can join the conference. If a **Guest PIN** has also been set, then the **PIN** will apply to the conference Host(s) only. For more information, see About PINs, Hosts and Guests.
allow_guests *	(Virtual Meeting Room and Virtual Auditorium only) Determines whether the conference will allow participants with Guest privileges. For more information, see About PINs, Hosts and Guests. • true — the conference has two types of participants: Hosts and Guests. The **pin** to be used by Hosts must be specified. A **guest_pin** can optionally be specified; if a **guest_pin** is not specified, Guests can join without a PIN. • false — all participants have Host privileges Default: false
guest_pin	(Virtual Meeting Room and Virtual Auditorium only) This optional field allows you to set a secure access code that must be entered by Guests before they can join the conference. For more information, see About PINs, Hosts and Guests.
alias_alias	The alias that, when received by Pexip Infinity, will cause it to route the call to this service (Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Test Call Service). The alias entered here must match exactly the alias as it is received by Pexip Infinity. Wildcards and regular expressions are not supported. In most cases, the alias received by Pexip Infinity will be the same as the alias that the conference participant dialed from their endpoint, but there are some exceptions, described in About aliases. You may also want to define multiple aliases for the same service to ensure that it can be accessed by devices and protocols that enforce specific alias formats — for more information, see Using multiple aliases to access the same service.
alias_description	An optional description of the alias. This is useful if you have more than one alias for a service.
tag	This optional field allows you to assign a unique identifier to this service, which you can then use to track use of the service. For more information, see Tracking usage with a service tag.
max_callrate_in	The maximum media bandwidth in bps that Pexip Infinity will receive from each individual participant dialed in to the service. Range 384000 to 4096000 bps. For more information see Restricting call bandwidth.

max_callrate_out	The maximum media bandwidth in bps that Pexip Infinity will send to each individual participant dialed in to the service. Range 384000 to 4096000 bps. For more information see Restricting call bandwidth.
call_type *	Allows you to limit the media content of the conference. For more information, see Controlling media capability. Valid values are: • video — main video plus presentation • video-only — main video only • audio — audio-only Default: video
host_view *	(Virtual Meeting Room and Virtual Auditorium only) The maximum number of other participants that each host participant will see, and the layout used to show them. For more information, see Selecting the layout seen by participants. Valid values are: • one_main_zero_pips — full-screen main speaker only • one_main_seven_pips — large main speaker and up to 7 other participants • one_main_twentyone_pips — main speaker and up to 21 other participants • two_mains_twentyone_pips — two main speakers and up to 21 other participants Default: one_main_seven_pips
guest_view *	(Virtual Auditorium only) The maximum number of Host participants that each guest participant will see, and the layout used to show them. (Guests will only see Hosts; they can hear but not see any of the other Guests.) For more information, see Selecting the layout seen by participants. Valid values are: • one_main_zero_pips — full-screen main speaker only • one_main_seven_pips — large main speaker and up to 7 other participants • one_main_twentyone_pips — main speaker and up to 21 other participants • two_mains_twentyone_pips — two main speakers and up to 21 other participants Default: one_main_seven_pips
force_presenter_into_main *	(Virtual Auditorium only) When a presentation is being shown, this option controls whether the main speaker position shows the presenter or the current speaker. For more information, see Selecting the layout seen by participants. • true — the Host sending the presentation stream will always hold the main video position • false — the main video position is voice-switched Default: false
enable_overlay_text *	(Virtual Meeting Room and Virtual Auditorium only) If enabled, the display name or alias of each main speaker is shown. For more information, see Showing the names of speakers. • true — speaker names are shown • false — speaker names are not shown Default: false
ivr_theme_name	The name of the theme to use with the service. If no theme is specified, the default Pexip theme is used. For more information, see Customizing images and voice prompts using themes.
participant_limit	This optional field allows you to limit the number of participants allowed to join this VMR. For more information see Limiting the number of participants.

primary_owner_email_address	(Virtual Meeting Room only) The email address of the owner of the VMR. VMR provisioning messages associated with this VMR will be sent to this address.
mute_all_guests *	(Virtual Auditorium only) Determines whether Guest participants will be muted when they first join the conference. • true — mute guests when they first join the conference • false — do not mute guests when they first join the conference
mssip_proxy_name	(Virtual Reception only) The name of the Lync / Skype for Business server to use to place the outgoing call. DNS is used if no server is specified.
match_string	(Virtual Reception only) An optional regular expression used to match against the alias entered by the caller into the Virtual Reception. If the entered alias does not match the expression, the Virtual Reception will not route the call. If this field is left blank, any entered alias is permitted. For more information, see Restricting or transforming the aliases entered into a Virtual Reception.
replace_string	(Virtual Reception only) An optional regular expression used to transform the alias entered by the caller into the Virtual Reception. (Only applies if a regex match string is also configured and the entered alias matches that regex.) Leave this field blank if you do not want to change the alias entered by the caller.
system_location_name	(Virtual Reception only) This is an optional field used in conjunction with the **mssip_proxy_name** setting. If specified, a Conferencing Node in this system location will perform the Lync / Skype for Business Conference ID lookup on the Lync / Skype for Business server. If a location is not specified, the IVR ingress node will perform the lookup.

* If this field is left blank, the default value will be used.

Multiple aliases

For services with more than one alias, you must add an additional record for each additional alias that repeats all fields except the **alias_alias** and **alias_description**:

```
name1,description1,pin1,allow_guests,guest_pin1,alias_alias1,alias_description1,...
name1,description1,pin1,allow_guests,guest_pin1,alias_alias2,alias_description2,...
```

Duplicates

If any records in the CSV file have the same **name** field, and any of the other fields apart from **alias_alias** and **alias_description** are different, only one service with that name will be created. This service will use the last record that was imported.

If any records in the CSV file have the same **name** as an existing service, the existing configuration will be overwritten by the imported service's configuration.

Restrictions

- The **name** field cannot be blank.
- All non-blank fields must contain valid data, for example the **pin** field must only contain digits.
- If non-ASCII characters are used, the file must be encoded as UTF-8 text.
- Fields with set values of **True** or **False** are not case-sensitive. All other fields are case-sensitive.

Examples

To import a Virtual Meeting Room called **alice** with a single alias of **meet.alice**, and a second Virtual Meeting Room called **bob** with aliases **meet.bob** and **meet.bobby**, you would create the following CSV file:

```
alice,,,,,meet.alice
bob,,,,,meet.bob
bob,,,,,meet.bobby
```

To import Virtual Meeting Rooms for Alice, Bob and Charlie that each have different Host and Guest PINs you would create the following CSV file:

```
alice,,1234,True,6789,meet.alice
bob,,4567,True,9876,meet.bob
charlie,,5432145,True,5556789,meet.charlie
```

Importing the CSV file

Virtual Meeting Rooms, Virtual Auditoriums and Virtual Receptions require separate CSV files.

To import the data in the CSV file to Pexip Infinity:

1. On the Pexip Infinity Administrator interface, go to either:
   - **Service Configuration > Virtual Meeting Rooms** or
   - **Service Configuration > Virtual Auditoriums** or
   - **Service Configuration > Virtual Receptions**.
2. Select **Import**.
3. Choose the CSV file to import and select **Save**.

Exporting existing service configuration

You can export all of your existing Virtual Meeting Room, Virtual Auditorium or Virtual Reception configuration data to a CSV file. This produces a CSV file in the same format as that used for importing configuration data (described above in Preparing the CSV file for import). The file includes a header row.

This feature exports all configuration except the Automatically Dialed Participants associated with each service.

To export the data:

1. On the Pexip Infinity Administrator interface, go to either:
   - **Service Configuration > Virtual Meeting Rooms** or
   - **Service Configuration > Virtual Auditoriums** or
   - **Service Configuration > Virtual Receptions**.
2. Select **Export**. This takes you to the **Export Configuration** page.
3. Select **Download**.
4. Follow your browser's prompts to save or open the file.

Bulk import/export of Automatically dialed participants (ADPs)

If you want to configure your Pexip Infinity platform with a large number of Automatically Dialed Participants (APDs) you can import the configuration from a CSV file.

Note that this process is separate to the Bulk import/export of service configuration.

You can also export all of your existing ADP configuration data to a CSV file. You may want to do this for backup purposes or to transfer configuration between, for example, a test system and a production system.

Preparing the CSV file for import

Header row

A header row in the CSV file is optional. If included, it should use exactly the same text for each field name as shown in the following section, although the order of the fields may vary.

Format

The data in the CSV file must use the following format, and fields must be in the same order as shown (unless you have used a header file):

```
alias,remote_display_name,description,protocol,dtmf_sequence,role,streaming,keep_conference_alive,call_type,routing,presentation_url,system_location_name,conference_name
```

where:

alias	The alias of the participant to dial when a conference starts.
remote_display_name	An optional friendly name for this participant. This may be used instead of the participant's alias in participant lists and as a text overlay in some layout configurations.
description	An optional description of the Automatically Dialed Participant.
protocol *	The protocol to use to place the outgoing call: • sip • h323 • rtmp • mssip (for calls to Microsoft Lync / Skype for Business) Default: sip The protocol will be ignored if the **routing** field is set to ***routing_rule***.
dtmf_sequence	An optional DTMF sequence to be transmitted after the call to the dialed participant starts. If one or more commas are used in the DTMF sequence (as a 2-second pause), the entire string within the field must be contained in double quotes.
role *	The level of privileges the participant has in the conference: • chair — the participant has Host privileges • guest — the participant has Guest privileges Default: guest
streaming *	Identifies the dialed participant as a streaming or recording device. • true — the participant is a streaming or recording device • false — the participant is not a streaming or recording device Default: false
keep_conference_alive *	Determines whether the conference continues when all other non-ADP participants have disconnected: • keep_conference_alive — the conference continues to run until this participant disconnects (applies to Hosts only). • keep_conference_alive_if_multiple — the conference continues to run as long as there are two or more "keep_conference_alive_if_multiple" participants and at least one of them is a Host. • keep_conference_alive_never — the conference terminates automatically if this is the only remaining participant. Default: keep_conference_alive_if_multiple

call_type *	Allows you to limit the media content of the call. The participant being called will not be able to escalate beyond the selected capability. For more information, see Controlling media capability. Valid values are: • video — main video plus presentation • video-only — main video only • audio — audio-only Default: video
routing *	Specifies how to route the call: • manual — uses the requested **protocol** and the defaults for the specified **system_location_name**. • routing_rule — routes the call according to the configured Call Routing Rules— this means that the dialed alias must match an outgoing Call Routing Rule for the call to be placed (using the protocols and call control systems etc. as configured for that rule). Default: manual
presentation_url	This additional parameter can be specified for RTMP calls to send the presentation stream to a separate RTMP destination.
system_location_name	The location of the Conferencing Node from which to place the call.
conference_name	The name of the Virtual Meeting Room or Virtual Auditorium from which this participant will be dialed automatically whenever a conference using that service starts.

* If this field is left blank, the default value will be used.

Duplicates

You can upload any number of records with the same **alias** field, as long as the **conference_name** field is different for each.

If any records in the CSV file have the same **alias** and **conference_name** fields, only one ADP for that conference that name will be created. This ADP will use the last record that was imported.

If any records in the CSV file have the same **alias** and **conference_name** as an existing ADP, the existing configuration will be overwritten by the imported ADP's configuration.

Restrictions

- The **alias** field cannot be blank.
- All non-blank fields must contain valid data, for example the **dtmf_sequence** field must only contain digits or commas.
- If non-ASCII characters are used, the file must be encoded as UTF-8 text.
- Fields with set values of **True** or **False** are not case-sensitive. All other fields are case-sensitive.

Examples

To import an ADP for Alice's endpoint (**alice@example.com**) that will be dialed in as a host when her VMR (**meet Alice**) is called, and as a guest when the sales team's Virtual Auditorium (**meet Sales**) is called, you would create the following CSV file:

```
alice@example.com,Alice Jones,,,,chair,,,,,,,meet Alice
alice@example.com,Alice Jones,,,,guest,,,,,,,meet Sales
```

Importing the CSV file

To import the data in the CSV file to Pexip Infinity:

1. On the Pexip Infinity Administrator interface, go to **Service Configuration > Automatically Dialed Participants**.
2. From the bottom of the page, select **Import**.
3. Choose the CSV file to import and select **Save**.

Exporting existing service configuration

You can export all of your existing ADP configuration data to a CSV file. This produces a CSV file in the same format as that used for importing configuration data (described above in Preparing the CSV file for import). The file includes a header row.

To export the data:

1. On the Pexip Infinity Administrator interface, go to **Service Configuration > Automatically Dialed Participants**.
2. From the bottom of the page, select **Export**. This takes you to the **Export Automatically Dialed Participant Configuration** page.
3. Select **Download**.
4. Follow your browser's prompts to save or open the file.

Bulk import/export of device aliases

Devices can register to Pexip Infinity using a permitted alias. If you want to configure your Pexip Infinity platform with a large number of device aliases you can import the configuration from a CSV file.

You can also export all of your existing device alias data to a CSV file. You may want to do this for backup purposes or to transfer configuration between, for example, a test system and a production system.

Preparing the CSV file for import

Header row

A header row in the CSV file is optional. If included, it should use exactly the same text for each field name as shown in the following section, although the order of the fields may vary.

Format

The data in the CSV file must use the following format, and fields must be in the same order as shown (unless you have used a header file):

```
alias,tag,description,username,password,primary_owner_email_address
```

where:

alias	The alias URI with which the device must register to Pexip Infinity. The alias must be an exact match; regular expressions are not supported.
tag	This optional field allows you to assign a unique identifier to this service, which you can then use to track use of the service. For more information, see Tracking usage with a service tag.
description	An optional description of the device.
username	The username with which to authenticate the device. The credentials are optional and the device is not challenged if no credentials are entered.
password	The password with which to authenticate the device. The credentials are optional and the device is not challenged if no credentials are entered. When passwords are exported to a CSV file, they will be encrypted. You can however re-import the encrypted form of the password and it will be saved as the original.
primary_owner_email_address	The email address of the owner of the device alias. Device provisioning messages associated with this alias will be sent to this address.

Duplicates

If any records in the CSV file have the same **alias** field, only one device alias with that name will be created. This device alias will use the last record that was imported.

If any records in the CSV file have the same **alias** as an existing device alias, the existing configuration will be overwritten by the imported device alias's configuration.

Restrictions

- The **alias** field cannot be blank.
- All non-blank fields must contain valid data.
- If non-ASCII characters are used, the file must be encoded as UTF-8 text.
- All fields are case-sensitive.

Examples

```
alias,tag,description,username,password,sync_tag,primary_owner_email_address
```

To create aliases that will allow Alice and Bob to register their devices to Pexip Infinity with a username and password, you would create the following CSV file:

```
alice@example.com,,,alice,1234
bob@example.com,,,bob,5678
```

Importing the CSV file

To import the data in the CSV file to Pexip Infinity:

1. On the Pexip Infinity Administrator interface, go to **Service Configuration > Device Aliases**.
2. From the bottom of the page, select **Import**.
3. Choose the CSV file to import and select **Save**.

Exporting existing service configuration

You can export all of your existing device alias configuration data to a CSV file. This produces a CSV file in the same format as that used for importing configuration data (described above in Preparing the CSV file for import). The file includes a header row.

To export the data:

1. On the Pexip Infinity Administrator interface, go to **Service Configuration > Device Aliases**.
2. From the bottom of the page, select **Export**. This takes you to the **Export Device Alias Configuration** page.
3. Select **Download**.
4. Follow your browser's prompts to save or open the file.

Pexip Infinity reference information

Glossary of Pexip Infinity terms

The following table contains a list of many of the terms used in Pexip Infinity documentation.

Term	Definition
Alias	The string that, when received by a **Conferencing Node**, will trigger the creation of a conference instance (or if one already exists, will cause the call to be routed to the appropriate conference). Each alias is associated with a service (Virtual Meeting Room, Virtual Auditorium or Virtual Reception), which defines the type and settings (such as PIN) for the conference that is created. In most cases, the alias received by the Conferencing Node will be the same as the alias that the conference participant dialed from their endpoint, but there are some exceptions (for more information see About aliases). Depending on the dial plan, multiple aliases can be used throughout a network to access the same service.
Automatically dialed participant	A participant who will have a call placed to them from a Virtual Meeting Room or Virtual Auditorium whenever a conference using that service starts. For more information, see Automatically dialing out to a participant from a conference.
Backplane	A link between the Management Node and a Conferencing Node, or between two Conferencing Nodes, used to transmit Pexip control messages. Backplanes between Conferencing Nodes also transmit conference audio, video, and data packets. All packets are secured through authentication and encryption designed to protect the privacy of the data. For more information, see Encryption methodologies. **Local backplanes** exist between Conferencing Nodes in a single location. **Geo backplanes** exist between Conferencing Nodes in different locations.
Conference instance	A conference with active participants that exists on one or more Conferencing Nodes. A unique conference instance is created when the first participant dials a Virtual Meeting Room or Virtual Auditorium alias and lasts until the last participant disconnects.
Conferencing Node	A virtual machine (VM) that provides the capacity for conferences, handling the media processing and call signaling. For more information, see Conferencing Nodes.
Core	One single physical processing unit. Intel Xeon E5 typically has 8 cores (10, 12 or more in newer versions)
Distributed conference	A conference instance that exists across two or more Conferencing Nodes. It can be **locally distributed**, **globally distributed**, or both: Locally distributed conferences exist across two or more Conferencing Nodes in the same location. Globally distributed conferences exist across two or more Conferencing Nodes in physically different locations. Locally and globally distributed conferences exist across two or more Conferencing Nodes in one location and at least one other Conferencing Node in a different location.
Endpoint	A device capable of participating in a conference. The endpoint's capabilities can vary from audio-only to full audio, video, and data sharing support.
Host	A conference participant who has privileges to control aspects of the conference. For more information, see About PINs, Hosts and Guests.
Host server	The physical hardware on which the virtual **Management Node** and **Conferencing Node**s reside. For more information, see Host servers.

Term	Definition
Hypervisor	An application that is used to create and manage virtual machines. For more information on the hypervisors supported by Pexip Infinity, see Supported hypervisors.
Infinity Connect	A suite of software clients that allows end users to connect to Pexip Infinity services from a web browser, an installable desktop client, or a mobile client. For more information, see Introduction to Infinity Connect.
IVR (Interactive Voice Response)	IVR technology allows participants to use a DTMF keypad to interact with Pexip Infinity services to: • select which Virtual Meeting Room or Virtual Auditorium they wish to join. For more information, see About the Virtual Reception IVR service. • enter a PIN, for those Virtual Meeting Rooms that have restricted access. For more information, see About PINs, Hosts and Guests.
Management Node	A virtual machine (VM) on which the **Pexip Infinity** software is installed. This machine hosts the Pexip Infinity Administrator interface. It is used to create one or more Conferencing Nodes and configure information about the conferences that can exist on those Conferencing Nodes.
NUMA node	The combination of a processor (consisting of one or more cores) and its associated memory.
Pexip Infinity	A virtual conferencing platform that encompasses a single Management Node and its associated Conferencing Node(s). Together they provide a coherent conferencing service to a single organization.
Pexip Infinity software	The files that are used to deploy the Management Node on to the physical virtual machine (VM) infrastructure.
Port	A connection can be a call or presentation from an endpoint to a Virtual Meeting Room or Virtual Auditorium, a backplane between Conferencing Nodes, or a call into or out of the Pexip Distributed Gateway. In this context, a **connection** is analogous to a **port**. For more information, see Pexip Infinity license installation and usage. Port can also refer to the virtual data connections used for network traffic between devices. For more information on the ports used by the Management Node and Conferencing Nodes to connect to other devices, see Pexip Infinity port usage and firewall guidance.
Processor	The hardware within a computer that carries out the basic computing functions. Can consist of multiple cores.
Proxy	In a globally distributed conference, one **Conferencing Node** in each location will act as the proxy for any other Conferencing Nodes in the same location. Each proxy will funnel traffic between physical locations. This will prevent full mesh bandwidth occupation, minimizing bandwidth consumption and ensuring optimal WAN utilization.
RAM	Also referred to as "memory". The hardware that stores data which is accessed by the processor core while executing programs.
Service tag	An optional identifier that an administrator can assign to a service, allowing them to track usage of the service via the administrator log. For more information, see Tracking usage with a service tag.
Socket (CPU)	The socket on the host server's motherboard where one processor is installed.
System location	A label that allows you to group Management Nodes and Conferencing Nodes together according to where they are physically located. For more information, see About system locations.
Thumbnail	A smaller window at the bottom of the main picture which displays the live video stream from a conference participant. Sometimes referred to as a PiP (Picture in Picture).
Virtual Machine (VM)	A software implementation of a computer, which runs on a host server and is implemented and managed using a hypervisor. The Management Node and Conferencing Nodes are virtual machines.
Virtual Auditorium	A meeting space that is optimized for use by a small number of Host participants and a large number of Guests. For more information, see About Virtual Auditoriums.
Virtual Meeting Room	A personal virtual meeting space. For more information, see About Virtual Meeting Rooms.
Virtual Reception	A central IVR service from which participants can select the Virtual Meeting Room or Virtual Auditorium they wish to join. For more information, see About the Virtual Reception IVR service.

Term	Definition
VM manager	An application that allows you to connect to one or more VMware vSphere ESXi Hypervisors (which manage host servers and their virtual machines). VM managers supported by version 14.1 of Pexip Infinity are vCenter Server and vSphere. • **vCenter Server** is an application that is used to manage one or more host servers running ESXi, through a single interface. • **vSphere** on the host is used when managing a single host server running ESXi. For more information see Using a VM manager.

Regular expression (regex) reference

Regular expressions can be used in conjunction with Pexip Infinity features including Call Routing Rules, Provisioning VMRs and devices from Active Directory via LDAP and searching the support and admin logs.

Pexip Infinity supports case-insensitive Perl-style regular expression patterns. The table below describes some of the special characters that are commonly used in regular expressions, and provides examples of how they can be used.

Character	Description	Example
Basic syntax		
non-special character	Matches the specified characters literally (providing they are not any of the regex special characters).	**abc** matches **abc**, **ABC**, **aBC** etc
(...)	Groups a set of matching characters together. Multiple groups can be specified. Each group can then be referenced in order (from left to right) using the characters **\1**, **\2**, etc. as part of a replace string.	See Search and replace examples below
\	Escapes a regular expression special character: **{}[]()^ $.|*+?** Also used to reference a group in a replace expression.	**ab\+** matches **ab+**
|	Matches either one expression or an alternative expression.	**.*@example\.(net|com)** matches against any URI for the domain example.com or the domain example.net
Wildcard and character matching		
.	Matches any single character.	**a.c** matches **aac**, **abc**, **azc**, **a2c**, **a$c** etc.
\d	Matches a decimal digit character (i.e. 0-9).	**a\d** matches **a1**, **a2**, **a3** etc. but not **aa**, **ab** etc
\D	Matches a non-digit character.	**a\D** matches **ab** but not **a1**, **a2**, **a3** etc
\s	Matches any whitespace character (space, tab, newline).	**ab\sd** matches **ab d** but not **abcd**, **abxd** etc
\S	Matches any non-whitespace character.	**ab\Sd** matches **abcd**, **abxd** etc. but not **ab d**
[...]	Matches the characters specified in the brackets. You can specify a range of characters by specifying the first and last characters in the range, separated by a hyphen. You cannot use other special characters within the [] — they will be taken literally.	**9[aeiou]** matches **9a**, **9e**, **9i** etc. and **9A**, **9E**, **9I** etc, but not **9b**, **9c**, **9B**, **9C** etc **9[a-z]** matches **9a**, **9b**, **9z** etc. and **9A**, **9B**, **9Z** etc. but not strings such as **91**, **99** or **9(** **[0-9#*]** matches any single E.164 character (digits 0-9, hash key or asterisk)
[^...]	Matches anything except the set of specified characters.	**[^a-z]** matches any non-alphabetical character **[^0-9#*]** matches anything other than an E.164 character

Character	Description	Example
Repetition factors		
*	Matches 0 or more repetitions of the previous character or expression.	**ab*c** matches **ac**, **abc**, **abbc**, but not **ab** or **abd**
+	Matches 1 or more repetitions of the previous character or expression.	**ab+c** matches **abc** and **abbc**, but not **ab**, **ac** or **abd** **[a-z.]+** matches any string containing the letters a to z, A to Z, or dots (periods)
?	Matches 0 or 1 repetitions of the previous character or expression.	**ab?c** matches **ac** and **abc**, but not **ab**, **abbc** or **abd**
.*	Matches against any sequence of characters.	**a.*** matches everything beginning with **a**
{n}	Matches n repetitions of the previous character or expression.	**ab{2}c** matches **abbc**, but not **abc** or **abbbc** **a\d{3}** matches **a123**and **a789**, but not **a12** or **456**
{n,m}	Matches n to m repetitions of the previous character or expression.	**ab{2,4}c** matches **abbc**, **abbbc** and **abbbbc**, but not **abc** or **abbbbbc**
{n,}	Matches at least n repetitions of the previous character or expression.	**ab{2,}c** matches **abbc**, **abbbc**, **abbbbc** etc, but not **abc**
Position matching		
^	Matches the beginning of a line.	**^meet\.(.*)** matches any string that starts with **meet.** and places the rest of the string into a group (which could be used in a replace string).
$	Matches the end of the line.	**.*\.com$** matches any string that ends in **.com**
(?!...)	Negative lookahead. Defines a subexpression that must not be present immediately after the current match position, for example **regex1 (?! regex2)** where a match is found if **regex1** matches and **regex2** does not match.	**(?!.*@example\.com$).*** matches any string that does not end with **@example.com** **(?!meet).*** matches any string that does not start with **meet** **meet(?!\.).*** matches any string that starts with **meet** providing it is not followed by a period
(?<!...)	Negative lookbehind. Defines a subexpression that must not be present immediately before the current match position, for example **(?<!regex1)regex2** where a match is found if **regex1** does not match and **regex2** matches.	**.*(?<!@)example\.com.*** matches any string containing **example.com** providing it is not preceded by **@**

Note that this is only a subset of the full range of expressions. For a full description of regular expression syntax see http://perldoc.perl.org/perlre.html. You can test regex expressions at sites such as https://regex101.com/.

Pattern matching examples

VMR naming patterns

If your videoconferencing rooms have an alias naming convention in the form oslo1@example.com, oslo2@example.com, newyork1@example.com etc, this could be matched with an expression such as:

[a-z]+\d@example\.com

Or, if the rooms are named 555 followed by exactly three digits, e.g. 555100@example.com and 555234@example.com, you could use the expression:

555\d{3}@example\.com

If your room aliases could follow either pattern, you could merge the two expressions like this:

([a-z]+\d|555\d{3})@example\.com

Matching an IP address

To match against an IP address, use:

^(([0-9]|[1-9][0-9]|1[0-9]{2}|2[0-4][0-9]|25[0-5])\.){3}([0-9]|[1-9][0-9]|1[0-9]{2}|2[0-4][0-9]|25[0-5])$

To match against somebody dialing an IP address (e.g. from an H.323 device or the Web App), **or** someone dialing a SIP URI in the format IP_address@example.com, use a **Destination alias regex match** of:

^((([0-9]|[1-9][0-9]|1[0-9]{2}|2[0-4][0-9]|25[0-5])\.){3}([0-9]|[1-9][0-9]|1[0-9]{2}|2[0-4][0-9]|25[0-5]))(@example\.com)?$

with a **Destination alias regex replace string** of: **\1**

and this combination of settings will ignore any @example.com element and pass just the IP address.

Search and replace examples

Replacing an alias domain

This example transforms any alias that ends in example.net into an alias that ends in example.com:

Match string:	**(.+)(@example\.net$)**
Replace string:	**\1@example.com**

This example builds on the previous example by transforming any alias that ends in example.com, example.net or example.co.uk into a common alias that ends in example.com:

Match string:	**(.+)(@example\.(com|net|co\.uk)$)**
Replace string:	**\1@example.com**

Strip leading 9

This example strips a leading 9 from any all-numeric (0-9) alias (e.g. for integrating with an ITSP phone gateway service):

Match string:	**9([0-9]+$)**
Replace string:	**\1**

If your environment includes a PSTN gateway or uses an ITSP (Internet telephony service provider), consider the potential for toll fraud if you have Call Routing Rules that can route calls to the PSTN gateway or ITSP, or if you allow conference participants to dial out to other participants via the PSTN gateway or ITSP. See PSTN gateways and toll fraud for more information.

Encryption methodologies

Pexip nodes

The backplane (the link between the Management Node and a Conferencing Node, or between two Conferencing Nodes) uses an IPsec transport with the following settings:

- 256-bit AES-CBC for encryption
- SHA 512 hashing for integrity checking
- a 4096 bit Diffie-Hellman modulus for key exchange.

No other ciphers, hashes or moduli are permitted.

These settings apply to both the initial channel set up for key exchange (ISAKMP) and the secondary channel over which application data is transported (ESP).

Endpoints

Encrypted connections between Pexip Infinity and endpoints use:

- AES 128-bit encryption for media
- TLS for SIP call control (for more information, see Managing TLS and trusted CA certificates)
- SRTP for SIP media
- H.235 for H.323 media

Infinity Connect (web/desktop/mobile) clients use:

- HTTPS TLS for signaling
- DTLS and SRTMP (encrypted RTMP) for media

Interoperability

We endeavor to make Pexip Infinity interoperable with all relevant standards-based devices. These tables list the devices and software versions with which Pexip Infinity is known to work, along with any known issues. Note however that these lists are not exhaustive - the absence of a device or specific version does not imply a lack of interoperability.

We welcome your feedback on any known issues with any of the mentioned devices or software versions, and we would also like to hear of any other devices that you have used with Pexip Infinity. Please send your comments to your Pexip authorized support representative.

For a list of more general system issues and how to resolve them, see Troubleshooting the Pexip Infinity platform.

Microsoft

Product	Version	Notes
Endpoints		
Lync Room System (LRS) Skype Room Systems (SRS) (project Rigel - e.g. Logitech SmartDock)	from first release	
Microsoft Surface Hub	from first release	
Software endpoints		
Skype for Business 2016 client	Windows and OSX	See **Received content can be slow to update**.
Skype for Business 2015 client	Windows	
Skype for Business Mobile	iOS/Android/Windows	
Lync 2013 desktop client	Windows and OSX	
Lync 2011 desktop client	OSX	
Lync 2010 desktop client	Windows	
Call control		
Skype for Business Server	2015	
Lync Server	2010 2013	

Polycom

Product	Version	Notes
Hardware endpoints		
Polycom HDX 4000/4500/6000/7000/8000/9000 series	2.5 and above	
Polycom VSX 3000/5000/6000/7000/8000 series	8.7.1 and above	
Polycom QDX 6000	4.0.2 and above	
Polycom RPX 200/400 Immersive Room*	2.6 and above	
Polycom OTX 300 Immersive Room*	5.1 and above	
Polycom TPX 306 Immersive Room*	3.1 and above	
HP Halo Polycom Rooms	3.0 and above	
Polycom RealPresence Group 300/500/700 series	4.0.2 and above	
Polycom RealPresence Immersive Studio Room*	5.1 and above	
Polycom RealPresence Centro	5.1 and above	
Polycom RealPresence Debut	1.1 and above	
Polycom RealPresence Trio	5.4 and above	
Polycom VVX 1/2/3/4/5/600 Phone	5.4 and above	
Software endpoints		
Polycom RealPresence Desktop & Mobile – Windows, Mac, iOS, Android	2.3.0 and above	
Polycom CMA Desktop	5.2.2 and above	
Infrastructure		
Polycom DMA 7000	6.1.2	
Polycom CMA 4000/5000	4.0 and above	
Polycom VBP 5300/6400	9.0 and above	
Polycom RMX/RPCS 800s/1500/1800/2000/4000	8.2.0 and above	

* Pexip Infinity can support the primary VTC in a multi-screen system. If all screens are required, Pexip Infinity can be a gateway into the current MCU for an end-to-end experience.

Cisco

Product	Version	Notes
Hardware endpoints (VTC)		
Cisco C series (C20, C40, C60, C90)	TC4.2.1 and above	
Cisco DX series (DX80)	10.2.3	

Product	Version	Notes
Cisco E20	TE4.1.1 and above	See Main video on a Cisco E20 freezes when content is received
Cisco EX series (EX60, EX90)	TC6.3 and above	
Cisco MX series (MX200, MX300, MX700, MX800)	TC5.0 and above CE8.0 and above	
Cisco MXP series (880 MXP, 990 MXP, 1000 MXP, 1500 MXP, 1700 MXP, 3000 MXP, 6000 MXP)	F8.3 and above	See A Cisco MXP intermittently puts a call on hold immediately after resuming it. and After upgrading to version 10 or later, some Cisco MXPs may no longer be able to connect over TLS to Pexip Infinity.
Cisco Edge 75/85/95 MXP series	F5.3 and above	
Cisco SX series VTC systems (SX10, SX20, SX80)	TC7.3.0 and above	
Cisco Video Phones 9951/9971	9.3 and above	
Cisco TX 9000 Immersive Room*	6.1 and above	
Cisco IX 5000 Immersive Room*	8.0 and above	
Cisco Tandberg T1000	E5.3.1	
Cisco Tandberg T150	L6.1	
Software endpoints		
Cisco Jabber/Movi desktop client	4.3 and above	
Cisco Jabber for Windows (via CUCM)	9.7 and above	
Webex CMR	31.0 and above	
Call control		
Cisco VCS Control Cisco VCS Expressway	X7.2.1 and above	
Cisco Unified Communications Manager (CUCM)	8.5 and above	
Infrastructure		
Cisco Codian 4200 series (4203, 4205, 4210, 4215, 4220)	4.3 and above	
Cisco Codian 4500 series (4501, 4505, 4510, 4520)	4.3 and above	
Cisco Codian 8400 series	4.5 and above	
Cisco Codian 8500 series	4.3, and above	
Cisco TelePresence Server (8700, 7010)	3.1	
Cisco TelePresence MSE8000 MCU	4.1 and above	

* Pexip Infinity can support the primary VTC in a multi-screen system. If all screens are required, Pexip Infinity can be a gateway into the current MCU for an end-to-end experience.

Other manufacturers

Product	Version	Notes
Hardware endpoints		
Aethra Vega X5, X7	10.1.24, 12.1.7	
AVer EVC series	1.08, 10.06, 10.10	
AVer HVC3300	3.12, 3.14	
Counterpath Bria	3.2.1, 3.5.5	
Huawei TEX0 series HD Endpoints	1.1, 1.2	
Huawei - VP9030-1080P	11.2	
LifeSize Express 200, 220	4.8 and above	
Lifesize Room 200, 220	4.8 and above	
Lifesize Team 200, 220	4.8 and above	
Lifesize Passport	4.8 and above	
LifeSize Icon 600	1.1.4 and above	
RadVision XT1000, XT4000, XT5000, XT240	8.3 and above	
RadVision VC240	2.5 and above	
Sony PCS-XG80	2.30 and above	
Starleaf	1.8.2	
Tely Labs	4.5.1 and above	
Yealink VC110	50.0 and above	
Yealink VC120	40.0 and above	
Yealink VC400 VTC	30.0 and above	
Yealink T49G video phone	51.0 and above	
Software endpoints		
LifeSize Softphone	8.1.12	
Linphone	3.7, 3.8	
Yealink VC Desktop	1.0 and above	
Infrastructure		
GNU Gatekeeper		
LifeSize Media Server	5.2 and above	
Magor SBC	102 and above	
RadVision ViaIP MCU	7.7.4 and above	
RadVision ViaIP 100 / 400 Gateway	5.7.2	
VidyoGateway	1.20.2, 3.2.0	
Zoom Room Connector	1.0.1	

Hypervisors

For more information, see Supported hypervisors.

Product	Version	Notes
Microsoft Hyper-V	Microsoft Hyper-V Server 2012 and later Windows Server 2012 and later	
VMware ESXi	5.5 and above	
VMware vCloud Director	5.5 and above	
KVM Open Source	Linux 3.10.0 or later QEMU 1.5.0 or later	
Citrix Xen Server	6.0 and above	
Xen Server Open Source	4.2 and above	

Public cloud providers

For more information, see Deploying Pexip Infinity on Microsoft Azure and Deploying Pexip Infinity on Amazon Web Services (AWS).

Product	Version	Notes
Azure (Management Node)	A2	
Azure (Conferencing Node)	D4_v2 or F8	
AWS (Management Node)	m4.large	
AWS (Conferencing Node)	c4.2xlarge	

Browser video support

For more information, see About the Infinity Connect Web App.

Product	Version	Notes
Google Chrome WebRTC (VP8/VP9/H.264)	43 and later	
Microsoft Edge ORTC	20.10532 or later	Earlier versions will connect over RTMP and use Flash video.
Mozilla Firefox	39 and later	
Microsoft Internet Explorer	10 and later	Requires Flash Player 11 and later ActiveX® plug-in, and must not be in Compatibility View.
Apple Safari (with Adobe Flash Player 11)	6 and later	(Mac OS X only) Requires Adobe Flash Player 11 and later plug-in.
Opera	23 and later	

Notes:

- RTMP media streaming is also supported in all browsers.
- Instant Messaging to and from Lync / Skype for Business clients is supported in all browsers.

Supported RFCs

Pexip Infinity supports the following RFCs:

- RFC 1889 RTP: A Transport Protocol for Real-time Applications
- RFC 2190 RTP Payload Format for H.263 Video Streams
- RFC 2429 RTP Payload Format for the 1998 Version of ITU-T Rec. H.263 Video (H.263+)
- RFC 2782 DNS RR for specifying the location of services (DNS SRV)
- RFC 2790 Host Resources MIB
- RFC 2976 The SIP INFO Method
- RFC 3261 SIP: Session Initiation Protocol
- RFC 3263 Locating SIP Servers
- RFC 3264 An Offer/Answer Model with SDP
- RFC 3550 RTP: A Transport Protocol for Real-Time Applications
- RFC 3581 Symmetric Response Routing
- RFC 3605 RTCP attribute in SDP
- RFC 3711 The Secure Real-time Transport Protocol (SRTP)
- RFC 3840 Indicating User Agent Capabilities in SIP
- RFC 3890 A Transport Independent Bandwidth Modifier for SDP
- RFC 3891 SIP "Replaces" Header
- RFC 3984 RTP Payload Format for H.264 Video
- RFC 4320 Actions Addressing Identified Issues with the Session Initiation Protocol's (SIP) Non-INVITE Transaction
- RFC 4321 Problems Identified Associated with the Session Initiation Protocol's (SIP) Non-INVITE Transaction
- RFC 4566 SDP: Session Description Protocol
- RFC 4568 SDP: Security Descriptions for Media Streams
- RFC 4574 The Session Description Protocol (SDP) Label Attribute
- RFC 4582 The Binary Floor Control Protocol
- RFC 4583 SDP Format for BFCP Streams
- RFC 4585 Extended RTP Profile for RTCP-Based Feedback
- RFC 4587 RTP Payload Format for H.261 Video Streams
- RFC 4629 RTP Payload Format for ITU-T Rec. H.263 Video
- RFC 4733 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 4796 The SDP Content Attribute
- RFC 5168 XML Schema for Media Control
- RFC 5245 Interactive Connectivity Establishment (ICE)
- RFC 5389 Session Traversal Utilities for NAT (STUN)
- RFC 5577 RTP Payload Format for ITU-T Recommendation G.722.1
- RFC 5763 Framework for Establishing a Secure Real-time Transport Protocol (SRTP) Security Context Using Datagram Transport Layer Security (DTLS)
- RFC 5766 Traversal Using Relays around NAT (TURN)
- RFC 6026 Correct Transaction Handling for 2xx Responses to Session Initiation Protocol (SIP) INVITE Requests
- RFC 6416 RTP Payload Format for MPEG-4 Audio/Visual Streams
- draft-ietf-bfcpbis-rfc4582bis-10.txt
- draft-ietf-bfcpbis-rfc4583bis-08.txt
- draft-ietf-avt-rtp-h264-params-01.txt
- draft-ietf-payload-rtp-opus-01.txt
- draft-ietf-payload-rtp-vp8-10.txt
- draft-ietf-mmusic-sdp-g723-g729-04.txt

Changing aspect ratios

Endpoints send and display video images and presentations in various aspect ratios, most commonly 16:9 and 4:3. Pexip Infinity will only send images and presentations in 16:9 or 4:3. If there is a difference between the aspect ratios of the sending and receiving endpoints, then the endpoint and/or Pexip Infinity may crop the image or add black vertical or horizontal borders (also known as letterboxing and pillarboxing). How and when aspect ratios are changed depends on a number of factors, but the general principles are described below.

Presentations

Pexip Infinity

In general, Pexip Infinity prefers to send presentations as a 16:9 video stream (typically 720p, or 1080p if it is enabled) for calls where the bandwidth is sufficient and the endpoint supports it. But if this is not feasible, all lower bandwidth resolutions have an aspect ratio of 4:3.

If a change of aspect ratio is required, Pexip Infinity will add black borders (in order to not remove anything important from the presentation). It will never crop presentations.

Endpoints

If the aspect ratio of the presentation image being sent from Pexip Infinity does not match the aspect ratio of the endpoint, the endpoint will typically add another border to make it fit. Generally endpoints will not crop the image.

Main video

Pexip Infinity

If required, Pexip Infinity will change the aspect ratio of main video by cropping the image (to avoid black bars in the thumbnail images).

Endpoints

If the aspect ratio of the main video image being sent from Pexip Infinity does not match the aspect ratio of the endpoint, the endpoint will typically add a border to make it fit. Generally endpoints will not crop the image.

PSTN gateways and toll fraud

If your environment includes a PSTN gateway or uses an ITSP (Internet telephony service provider), consider the potential for toll fraud if you have Call Routing Rules that can route calls to the PSTN gateway or ITSP, or if you allow conference participants to dial out to other participants via the PSTN gateway or ITSP.

As you might intentionally want to allow users to route calls via the PSTN gateway or ITSP (and thus incur toll charges), we recommend that you use a suitable call control solution such as a Cisco VCS to configure an appropriate dial plan and authentication mechanism for your network.

The ways in which Pexip Infinity calls may be routed via a PSTN gateway or ITSP are described below.

Pexip Distributed Gateway calls

If your dial plan allows Pexip Distributed Gateway calls and has a Call Routing Rule which, for example, matched 9.* and routed the call via the PSTN gateway or ITSP, this would allow anyone who could route a call to the Pexip Infinity platform to then send a call via the PSTN gateway or ITSP.

Virtual Receptions

If you have configured Virtual Receptions and also have Call Routing Rules that match numeric aliases (such as 9.*), then anyone that can reach the Virtual Reception could match the Call Routing Rule and potentially route their call via the PSTN gateway or ITSP.

Thus, if any call control rules are in place to restrict who may dial numbers which correspond with numeric Call Routing Rules, then the same restrictions should also be placed on who may call any Virtual Receptions.

Manually dialing out to a participant from a conference

Pexip Infinity allows you to manually dial out to participants from a conference, on an ad-hoc basis.

This means that conference participants using Infinity Connect clients could place outbound calls via the PSTN gateway or ITSP. Note that these types of calls may dial out directly to the destination alias or they may use Call Routing Rules.

In these circumstances, to reduce (but not eliminate) the risk of toll fraud, we recommend that you use:

- PIN-protected conferences
- an appropriately configured reverse proxy that only allows authenticated connections from Infinity Connect clients.

Information sent when usage reporting is enabled

The Pexip Infinity platform includes a feature that allows you to send anonymized statistical information to Pexip for analysis. For more information on this feature and how to enable or disable it, see Automatically sending usage statistics.

When this feature is enabled, the following usage statistics are sent:

Information sent
Common
The version of Pexip Infinity software running on the Management Node.
The Pexip Infinity software license ID.
Public IP address of the Management Node from which the report has been sent.
The name of the Management Node.
Call
Reason the call was disconnected.
Whether the call was to a Virtual Meeting Room, Virtual Auditorium, Virtual Reception, or Pexip Distributed Gateway.
Number of seconds this call leg was alive.
Vendor of the endpoint that placed the call.
Protocol used by the endpoint (SIP, H.323, WebRTC, etc).
Maximum bandwidth signaled.
Whether the participant was a Guest or Host.
IP address of the Conferencing Node that handled the media for the call.
IP address of Conferencing Node that handled the signaling for the call.
Whether the media stream was encrypted.
Unique identifier of this call (UUID).
Unique identifier of the conference instance (UUID).
The time at which the call was connected.
The time at which the call was disconnected.
Identifier of the system location of the Conferencing Node to which the call was connected.
Whether the call was incoming or outgoing.
Audio/Video/Presentation (statistics for each type of stream, where present)
Whether the stream is audio, video or presentation.
The format used by the Conferencing Node to encode the media stream being sent to the endpoint.
The quantity of data sent to the endpoint.

Information sent
The display resolution of the image sent to the endpoint.
The total quantity of packets sent to the endpoint
The total quantity of packets sent from the Conferencing Node but not received by the endpoint.
The format used by the Conferencing Node to decode the media stream being sent from the endpoint.
The quantity of data received from the endpoint.
The display resolution of the image received from the endpoint.
The total quantity of packets received by the Conferencing Node from the endpoint.
The total quantity of packets sent from the endpoint but not received by the Conferencing Node.
Conference
Cryptographic hash of the name of the Virtual Meeting Room, Virtual Auditorium, Virtual Reception or Pexip Distributed Gateway that was used.
The date and time that the first participant connected to the service.
The date and time that the last participant's call ended.
Number of seconds conference was active.
Whether the call was to a Virtual Meeting Room, Virtual Auditorium, Virtual Reception, or Pexip Distributed Gateway.
Total number of participants who connected to this conference instance.
The number of chat messages exchanged on this conference.
Platform (sent once a day for each system location)
Numeric identifier of the system location.
Number of nodes in that system location.
Configuration
Whether RTMP is enabled.
Whether SIP (TCP and TLS) is enabled.
Whether SIP TLS certificate verification mode is set to ***On*** or ***Off***.
Whether WebRTC is enabled.
The URL to which OCSP requests will be sent.
The start value for the range of ports that all Conferencing Nodes use to send media.
The end value for the range of ports that all Conferencing Nodes use to send media.
The start value for the range of ports that all Conferencing Nodes use to send signaling.
The end value for the range of ports that all Conferencing Nodes use to send signaling.
Whether outbound calls are enabled.
Whether SSH is enabled.
Whether OCSP will be used when checking the validity of TLS certificates.
The DSCP value for management traffic sent from the Management Node and Conferencing Nodes.
The number of minutes a browser session may remain idle before the user is logged out of the Pexip Infinity Administrator interface.

Information sent
The URL to which incident reports will be sent (if enabled).
Whether HTTP access for external systems is enabled.
Whether H.323 is enabled.
Whether incident reporting is enabled.
Whether support for Pexip Infinity Connect and Mobile App is enabled.
The number of configured devices, whether or not they are registered.
The number of configured services (Virtual Meeting Rooms + Virtual Auditoriums + Virtual Receptions).
Whether chat messages are enabled.
Whether SIP UDP is enabled.
Registration
The number of users registered over SIP, H.323 and WebRTC.
Infinity Connect (when usage reporting is enabled on the client)
Indicates an attempt to place a call.
Indicates the call was connected.
Unique call identifier.
Properties of the type of client running Infinity Connect.
IP address of the client.
Width of the application window.
Height of the application window.
Screen width (maximum size available).
Screen height (maximum size available).
The bit depth of the color palette for displaying images.
The color resolution (in bits per pixel) of the screen.
Client/user language.
Whether the participant had audio-only media enabled.
Whether the participant had video (and audio) media enabled.
The requested connection bandwidth.
Indicates the participant initiated screen sharing / presenting.
Number of slides uploaded (when presenting).
Dial out participant role.
Dial out participant protocol.

Using Microsoft Lync / Skype for Business with Pexip Infinity

Pexip Infinity allows Microsoft Lync and Skype for Business* users to meet with other people regardless of the system they are using – Lync / Skype for Business, web browsers or traditional video conferencing systems. All participants can enjoy wideband audio, high definition video and cross-platform presentation sharing.

Pexip Infinity can be integrated with Lync/SfB as part of an existing, on-premises Lync/SfB environment inside an enterprise network, or as a standalone Pexip environment deployed in a public DMZ, leveraging direct federation with remote Lync/SfB environments. It enables full interoperability between Microsoft's H.264 SVC/RTV/RDP and H.263, H.264, VP8 (WebRTC) and BFCP/H.239 for truly seamless video and content sharing in any-to-any configurations, such as multiparty conferences.

In addition to enabling Lync/SfB participants to join conferences hosted on Pexip Infinity, Pexip Infinity can act as a gateway between Lync/SfB and standards-based endpoints. This enables Lync/SfB clients to receive and initiate point-to-point calls with H.323/SIP endpoints and registered Infinity Connect clients, and invite those devices into a Lync/SfB meeting while retaining the native meeting experience on each device.

* Note that where this documentation refers to "Lync/SfB", it represents both Microsoft Lync and Skype for Business unless explicitly stated otherwise.

Architecture options

Pexip Infinity can be integrated with Microsoft Lync in two different ways:

- As part of an existing, on-premises Lync environment inside an enterprise network, referred to as **on-prem deployment**.

  To integrate Pexip Infinity with an existing, on-premises Lync/SfB environment, one or more SIP domains are statically routed from the Lync/SfB environment towards one or more Pexip Infinity Conferencing Nodes. Then, when a Lync/SfB user dials a conference alias, such as meet.john@example.com, or the alias of a standards-based endpoint, the user is placed into the appropriate Pexip-hosted conference. The Lync/SfB user can also pin one or more such aliases to their contact list for easy access later.
- As a standalone Pexip environment deployed in a public DMZ, referred to as **public DMZ deployment**.

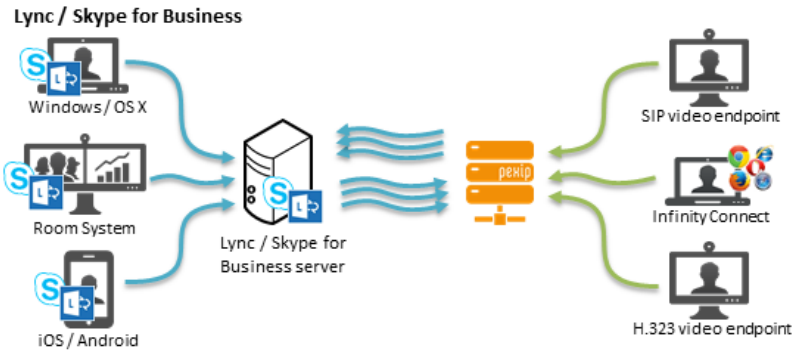
  As Pexip Infinity supports Lync/SfB natively, it can be deployed to enable Lync/SfB interoperability without having any existing, on-prem Lync/SfB infrastructure. In such a deployment, Pexip Infinity can federate directly with remote Lync/SfB environments (on-prem environments as well as Lync/SfB Online/Office 365), without the need for a local Lync/SfB environment.

You will typically choose one of the above two methods, depending on requirements and preference.

Pexip Infinity as a Lync gateway

Pexip Infinity can act as a gateway between Lync/SfB and standards-based endpoints. This enables Lync/SfB clients to:

- invite H.323/SIP endpoints and registered Infinity Connect clients into a Lync/SfB meeting
- use the Pexip Distributed Gateway service to route incoming calls directly into an ad hoc or scheduled Lync/SfB meeting
- when dialed into a Pexip VMR conference, invite other Lync/SfB or external contacts into that same Pexip



VMR (this creates a new Lync/SfB meeting which is merged with the existing Pexip VMR)
- receive and initiate person-to-person calls with standards-based devices.

More information

For full information on using Microsoft Lync with Pexip Infinity, see Pexip Infinity and Microsoft Lync Deployment Guide.

When is a reverse proxy, TURN server or STUN server required?

When your Conferencing Nodes are privately addressed, you will need to use a reverse proxy and a TURN server to allow external endpoints such as Infinity Connect and Lync / Skype for Business clients to access your Pexip Infinity services. A TURN server can also act as a STUN server, however, in some Pexip Infinity deployment scenarios where the TURN server is deployed inside your enterprise firewall, you may need to configure a separate, external STUN server.

When connecting to a privately-addressed Conferencing Node, Infinity Connect WebRTC clients that are behind a NAT may also use a STUN server to find out their public NAT address.

The following table shows when each of these devices needs to be deployed. When used, they must be publicly accessible, and routable from your Conferencing Nodes.

External endpoint / client	Conferencing Node addresses	Reverse proxy	TURN server	STUN server (for Conferencing Nodes)	STUN server (for WebRTC clients behind NAT)
Infinity Connect WebRTC clients	Private (on-premises)	✓	✓	✓ (if the TURN server is inside the firewall)	✓
Lync / Skype for Business clients*	Private (on-premises)	-	✓	✓ (if the TURN server is inside the firewall)	✓
Any endpoint / client	Publicly reachable — either directly or via static NAT	-	-	-	-

* Also requires a Lync / Skype for Business Edge server when Conferencing Nodes are privately addressed

Note that you may also need to deploy a reverse proxy if you want to host multiple Infinity Connect Web App customizations.

Pexip Infinity diagnostics

For information about how to monitor and troubleshoot Pexip Infinity, see:

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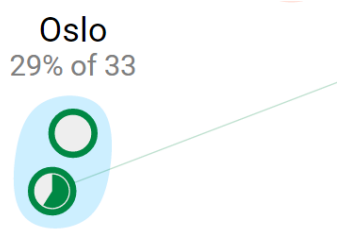
Viewing live platform status

For a high-level real-time overview of your Pexip Infinity deployment, go to **Status > Live View**. The interactive graph on this page shows you at a glance the locations and Conferencing Nodes that are currently deployed, and the available capacity and current load on each. It also shows the conferences currently taking place, and the nodes on which they are being hosted. You can view specific information on each location, node and conference, by hovering over each item.

Viewing location status

Each location is represented by one or more Conferencing Nodes against a colored background. A different color is used for each location. The label above gives the name of the location and a summary of the percentage of available HD ports across all Conferencing Node in that location that are currently in use.

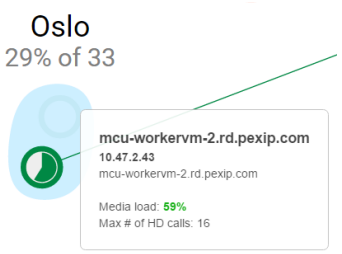
The example below shows the **Oslo** location, which is current using 29% of the 33 available HD ports:



Viewing Conferencing Node status

Within each location, each Conferencing Node is represented by a separate icon, described below. Hovering over each icon will provide information regarding that node's hostname, IP address, current load and estimated capacity (in terms of the estimated maximum number of HD calls the node can handle). Hovering over a node icon will also highlight the all the conferences for which that node is handling the media.

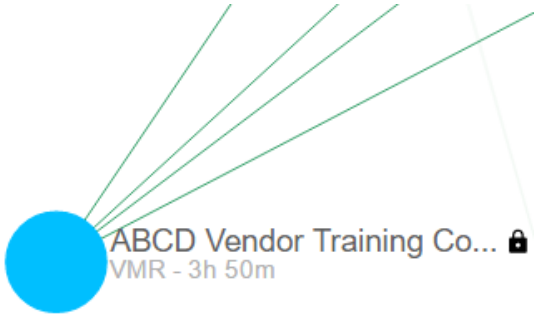
The example below shows the information that is available for one of the two Conferencing Nodes in the **Oslo** location:



Viewing conference status

Conferences are represented by blue circle icons, described below. Hovering over the icon will provide details of the conference, including what type of conference it is, how long it has been in progress, and the list of participants. Hovering over a conference icon will also highlight all the Conferencing Nodes that are handling media for that conference.

The example below shows the information that is available for the conference being hosted in the **ABCD Vendor Training Conference** Virtual Meeting Room:



Key to icons

Icon	Meaning
Conferencing Nodes *	
	A cloud bursting node which is currently off.
(Pulsating)	A cloud bursting node which is currently starting or stopping.
(Pulsating)	A node that hasn't reported for at least two minutes. (Nodes should report once each minute.)

Icon	Meaning
(Pulsating)	A node that is in maintenance mode.
	A node which is waiting to be upgraded. When you begin an upgrade of your platform, all nodes will have this status but will still be able to handle calls. After each node is upgraded in turn, it will return to normal operation.
(Pulsating)	A node which is currently upgrading.
Conferences **	A node during normal operation. The amount of green fill within the circle indicates the current media load (in terms of percentage of estimated HD ports in use), so an unused node will be white and fully loaded node will be filled entirely green.
	Conference being held in a Virtual Meeting Room.
	Conference being held in a Virtual Auditorium.

Icon	Meaning
	Call to a Virtual Reception.
	Gateway call.
	Call to the Test Call Service.
	Conference is locked.
Participants	
	The participant is currently presenting content.
	Streaming participant.
	API participant.

* For Conferencing Nodes, the size of each icon is relative to its total capacity.

** For conferences, the size of each icon is relative to the number of participants.

Service status

Viewing all aliases

To view a list of all existing aliases for all services (Virtual Meeting Rooms, Virtual Auditoriums and Virtual Receptions), go to **Service Configuration > Aliases**.

From here you can:

- Sort the list of aliases alphabetically by **Alias** or by **Service** name.
- **Search** for a particular alias.
- Edit an alias. To do this, click on the alias. This takes you to the **Change Alias** page. From here you can edit details of the alias, including changing the service with which it is associated.
- Add a new alias. To do this, click **Add Alias**. You will be asked to select the service with which this alias is to be associated.

Viewing current conference status

To see a list of all the conferences currently in progress, go to **Status > Conferences**.

To view historical information on conferences after they have finished, see Viewing historical information about conferences.

The pages showing conference status information do not refresh automatically — you must refresh them manually. A "**Conference instance no longer active**" message indicates that the conference whose details you were viewing has now finished.

Each conference has the following information available:

Field	Description
Service name	The name of the Virtual Meeting Room, Virtual Auditorium, Virtual Reception, Test Call Service or Call Routing Rule. For Pexip Distributed Gateway calls, the rule name is followed by a unique identifier to distinguish between separate calls. For Virtual Receptions and Test Call Services, if there are multiple concurrent users of that service you will see a single instance of that service (rather than one instance per participant, as all participants are using the same service even though they cannot see or hear each other). Click on the service name to view more information.
Duration	For Virtual Meeting Rooms and Virtual Auditoriums: the length of time since the first participant joined the conference. For Virtual Receptions and Test Call Services: the length of time that this instance of the service has been in continuous use (note that this could involve more than one participant if their usage overlapped). For Pexip Distributed Gateway calls: the length of time since the call was received by the Pexip Distributed Gateway.
Participant count	The number of participants currently in the conference or using the service.
Service type *	The type of conference, e.g. Virtual Meeting Room.
Service tag *	The unique identifier that an administrator has assigned to this service. If this field is blank, no tag has been assigned. For more information, see Tracking usage with a service tag.
Is locked *	Indicates whether a conference has been locked to prevent further participants from joining. For more information, see Locking a conference and allowing participants to join a locked conference.

* Only displayed when you have selected an individual conference to view.

To view more information about the conference, click on the service name. You will then see 3 tabs for the selected conference: Participants, Backplanes and Graph.

Participants

The **Participants** section lists all the participants that are currently in the conference.

For more details about a particular participant, including media stream statistics, click on the **Participant alias**. This takes you to the Participant details page.

Field	Description
Participant alias	The name of the user or the registered alias of the endpoint.
Duration	The length of time since this participant joined the conference or accessed the service.
Display name	The name that has been configured on the participant's endpoint.
System location	The system location of the Conferencing Node to which the endpoint is connected.
Signaling node	The IP address of the Conferencing Node to which the endpoint is connected. This Conferencing Node will be handling the call signaling but may or may not be handling the media (for more information, see Handling of media and signaling in locally distributed conferences).
Media node	The IP address of the Conferencing Node that is handling the call media for this endpoint (for more information, see Handling of media and signaling in locally distributed conferences).
Role	**Host** indicates that either: • the conference has no PINs configured (in which case all participants will have a role of **Host**) • the participant accessed the conference using the **Host PIN**. **Guest** indicates that the participant accessed the conference using the **Guest PIN**. **Unknown** indicates that either: • the participant is at the PIN entry screen and has not yet successfully entered a PIN • the participant is connected via a Lync / Skype for Business meeting (these participants will appear in the Infinity Connect participant list with a role of **External**). For more information, see About PINs, Hosts and Guests.
Is presenting	Indicates whether the endpoint is currently sending a presentation stream.
Is muted	Indicates whether the endpoint's audio has been muted (using an Infinity Connect client, the Administrator interface, or by a third party using the Pexip API).
On hold	Indicates whether the endpoint has been put on hold (usually by the endpoint user).
Streaming or recording device	Indicates if the participant is a streaming or recording device.

Backplanes

The **Backplanes** section provides information about the media streams being transmitted between Conferencing Nodes for the selected conference. Backplane links are unidirectional, so for a conference involving two Conferencing Nodes there will be two backplane links: one from node A to node B, and another from node B to node A.

Field	Description
Media node	The IP address of the Conferencing Node that is transmitting media. For details about the media streams being sent over a particular backplane link, click on the Media node's IP address.
Remote media node	The IP address of the Conferencing Node that is receiving media. Note that the remote media node of a merged Lync/SfB meeting is identified by the address of the Lync/SfB client that initiated the Lync/SfB meeting.
System location *	The system location of the Conferencing Node that is transmitting media.
Duration	The length of time since the connection was established.

Field	Description
Backplane type	**Geographic** indicates that the two Conferencing Nodes are in different system locations. **Local** indicates that the two Conferencing Nodes are in the same system location.

* Only displayed when you have selected an individual media node to view.

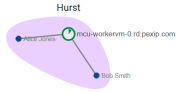
Backplane media streams

Media stream details are displayed when you have selected an individual node to view.

Field	Description
Type	Indicates whether the information is for an **Audio**, **Video**, or **Presentation** stream.
Start time	The time that the media stream started.
Tx codec	The format used by the transmitting Conferencing Node to encode and decode the media stream being transmitted.
Tx bitrate (kbps)	The quantity of data currently being sent from the transmitting Conferencing Node to the recipient Conferencing Node for this particular media stream.
Tx resolution	The display resolution of the image being sent from the transmitting Conferencing Node.
Tx packets sent	The total quantity of packets sent from the transmitting Conferencing Node to the recipient Conferencing Node since the start of the conference.
Tx packets lost	The total quantity of packets sent from the transmitting Conferencing Node but not received by the recipient Conferencing Node.
Tx jitter (ms)	The variation in the expected periodic arrival of packets being sent from the transmitting Conferencing Node to the recipient Conferencing Node, in milliseconds.

Graph

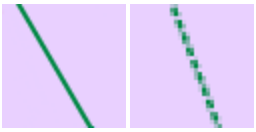
This section displays a graphical view of the connections for this conference, as follows:



Colored areas: each colored area highlights a system location and shows the Conferencing Nodes and endpoint connections within that location. A different color is used for each location.

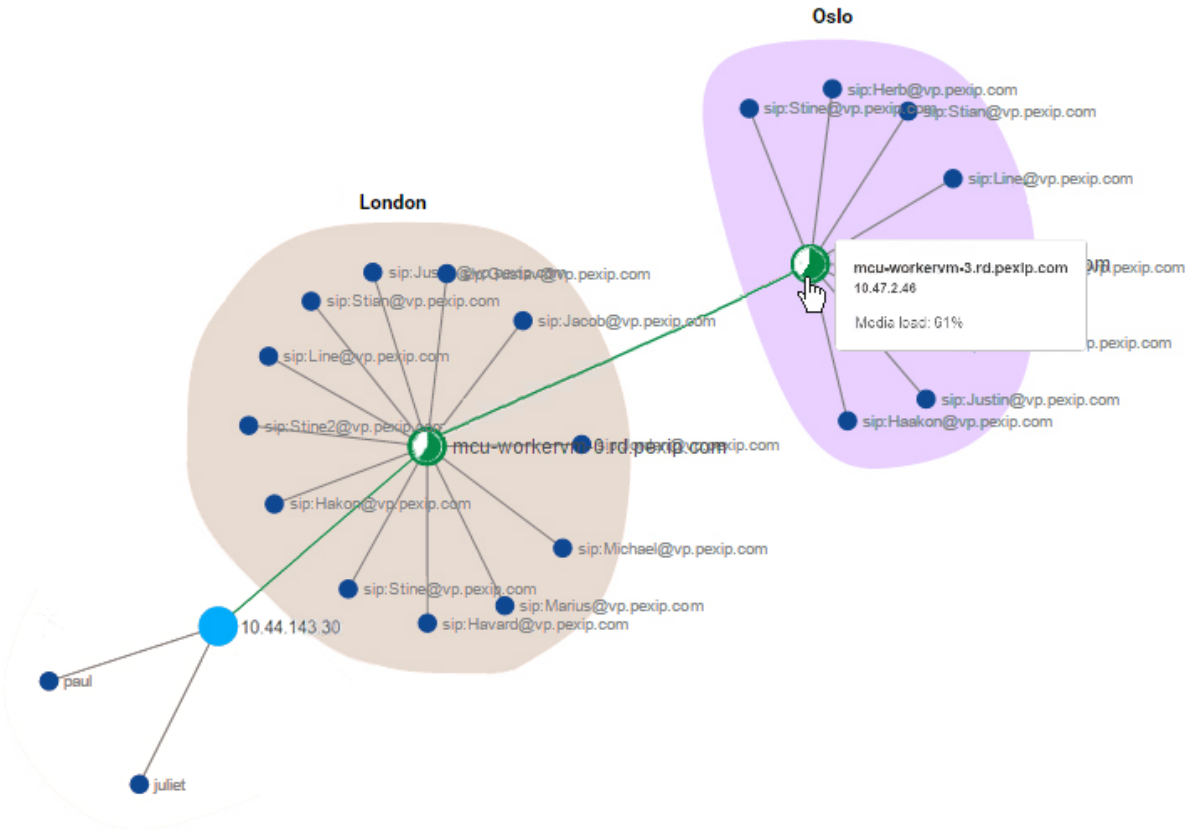
Small dark blue dots: all participant endpoints. Some may have an icon next to their name, as follows:

- for Infinity Connect control-only participants
- for participants who are currently presenting content
- for streaming participants.

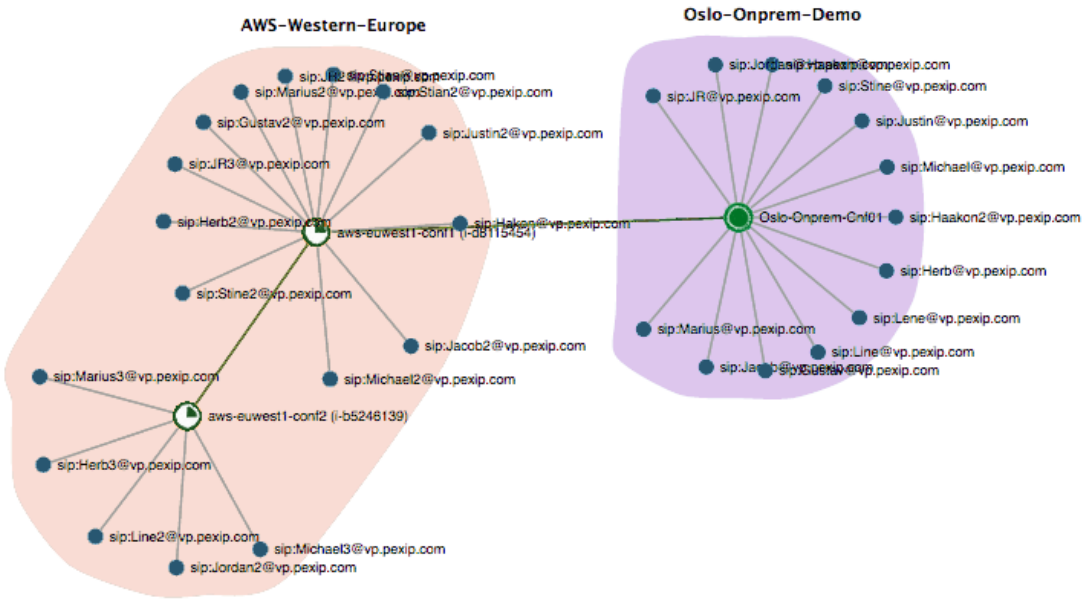
	Large green and white circles: the Conferencing Nodes to which the endpoints are connected. The amount of green fill within the circle indicates the current media load (in terms of percentage of estimated HD ports in use), so an unused node will be white and fully loaded node will be filled entirely green.
	Large pale blue dots: an external node (e.g. a Lync / Skype for Business meeting).
	**Green lines**: backplane links between Conferencing Nodes, or links to external nodes. These become **dashed green lines** if packet loss is greater than 1%.
	Gray lines: connections between an endpoint and a Conferencing Node. These become **dashed gray lines** if packet loss is greater than 1%.
	Red dashed lines: any connections with packet loss greater than 2%.
	Padlock: a padlock in the top right of the screen indicates that the conference is locked.

When viewing the graph you can:

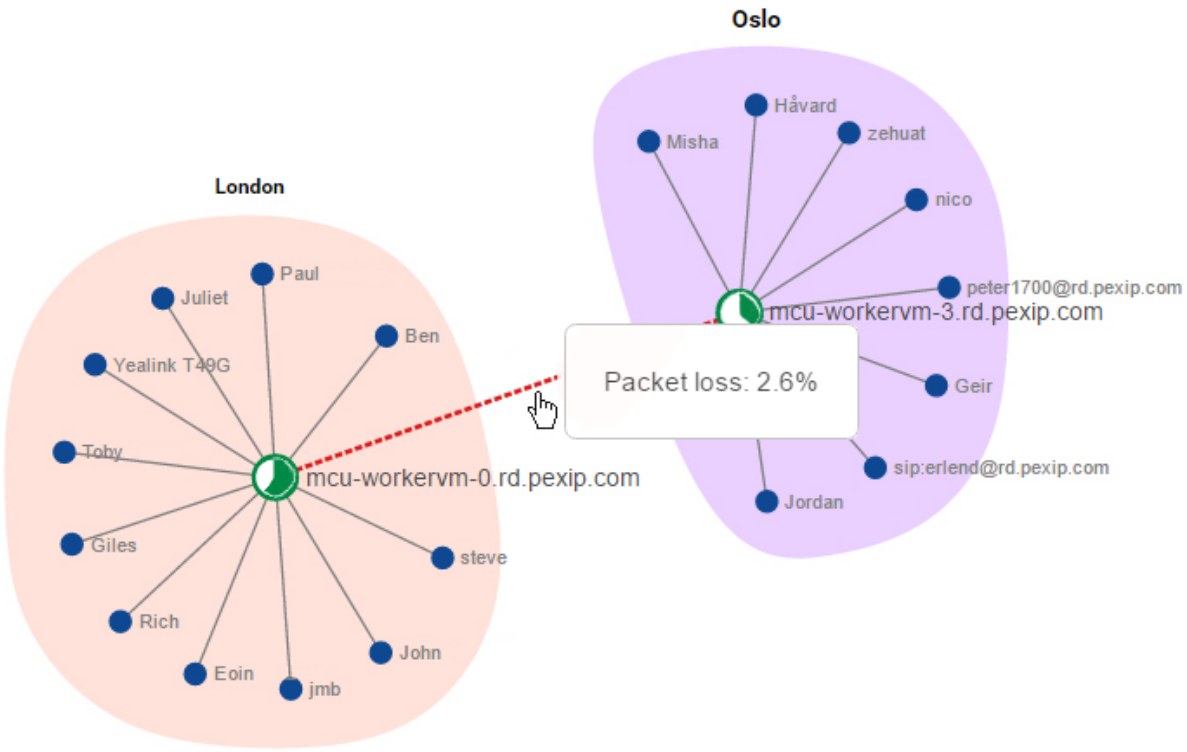
- view information on the % packet loss by hovering over a connection.
- view summary details of a particular participant or node by hovering over its icon.
- view full details of a participant or Conferencing Node by double-clicking on its icon.
- click within the graph to use your mouse to pan and zoom.



Example graph showing endpoints and a Lync / Skype for Business meeting connected to two Conferencing Nodes, where each Conferencing Node is in a separate location.



Example graph showing a system location containing two Conferencing Nodes that are hosted in AWS and are being used as an overflow location from an on-premises location containing one Conferencing Node.



Example graph showing packet loss on a backplane between two Conferencing Nodes in separate locations.

Viewing participant status

To see a list of all the current conference participants across all Pexip Infinity services, go to **Status > Participants**. This shows a list of all participants; to view a particular participant's details, click on the **Participant alias**.

When viewing a participant's details you can also use the controls at the bottom of the page to:

- **Mute** the participant
- **Disconnect** the participant from the conference
- **Transfer** the participant to another conference

To view historical information on participants after a conference has finished, see Viewing historical information about participants.

Participant details

The following information about each participant is available:

Field	Description
Participant alias	The name of the user or the registered alias of the endpoint. Click on the participant alias to view detailed information about the call.
Service name	The name of the Virtual Meeting Room, Virtual Auditorium, Virtual Reception, Test Call Service or Call Routing Rule. For Pexip Distributed Gateway calls, the rule name is followed by a unique identifier to distinguish between separate calls.
Connect time *	The date and time that signaling was established between the participant's endpoint and Pexip Infinity.
Duration	The length of time since this participant joined the conference or accessed the service.
Display name	The name that has been configured on the participant's endpoint.

Field	Description
Destination alias *	For participants that have dialed in to the conference or service themselves, this is the alias that they dialed. For participants that have been dialed out to manually or automatically from a Virtual Meeting Room or Virtual Auditorium, this is the alias of the endpoint that was dialed.
System location	The system location of the Conferencing Node to which the endpoint is connected.
Signaling node	The IP address of the Conferencing Node to which the endpoint is connected. This Conferencing Node will be handling the call signaling but may or may not be handling the media (for more information, see Handling of media and signaling in locally distributed conferences).
Media node	The IP address of the Conferencing Node that is handling the call media for this endpoint (for more information, see Handling of media and signaling in locally distributed conferences).
Service type *	**Virtual Meeting Room**, **Virtual Auditorium**, **Virtual Reception**, or **Gateway**: the participant has successfully accessed the service indicated. **PIN collection IVR**: the participant is currently accessing the Interactive Voice Response screens (where they are asked to enter a valid PIN). **Waiting for Host**: the participant is being shown a holding screen while they wait for a conference Host to join. **Insufficient Capacity Screen**: the participant is being shown a holding screen indicating that they cannot join the conference due to a lack of capacity on Pexip Infinity, or because the service's participant limit has been reached. **Insufficient Licenses Screen**: the participant is being shown a holding screen indicating that they cannot join the conference due to a lack of available licenses on Pexip Infinity. For more information, see Insufficient licenses. **Invalid License Screen**: the participant is being shown a holding screen indicating that they cannot join the conference because there are no valid licenses available on Pexip Infinity. For more information, see Invalid license.
Role	**Host** indicates that either: • the conference has no PINs configured (in which case all participants will have a role of **Host**) • the participant accessed the conference using the **Host PIN**. **Guest** indicates that the participant accessed the conference using the **Guest PIN**. **Unknown** indicates that either: • the participant is at the PIN entry screen and has not yet successfully entered a PIN • the participant is connected via a Lync / Skype for Business meeting (these participants will appear in the Infinity Connect participant list with a role of **External**). For more information, see About PINs, Hosts and Guests.
License count *	The number of licenses consumed by this participant. Media participants consume 1 license but API-only participants (e.g. Infinity Connect users who are not sending media) do not consume a license.
License type *	The type of license used, either "Port" (audio/video participant), "Audio" (audio-only participant — only applies if the system has audio licenses installed, otherwise port licenses are used for audio-only calls) or "None" (presentation or control only).
Is presentation stream supported *	Indicates whether the endpoint is able to support a separate media stream for presentations negotiated by H.239 or BFCP.
Is presenting	Indicates whether the endpoint is currently sending a presentation stream.
Is muted	Indicates whether the endpoint's audio has been muted (using an Infinity Connect client, the Administrator interface, or by a third party using the Pexip API).
On hold *	Indicates whether the endpoint has been put on hold (usually by the endpoint user).
Streaming or recording device *	Indicates if the participant is a streaming or recording device.

Field	Description
Protocol *	The communication protocol being used by the endpoint.
Call direction *	**In**: the call was placed by an external endpoint and received by Pexip Infinity. **Out**: the call was placed by Pexip Infinity to an endpoint or other device.
Bandwidth (kbps) *	The maximum bandwidth, in kbps, negotiated for use between the Conferencing Node and the endpoint. Actual bandwidth used is shown in the Media streams section (**Tx bitrate** and **Rx bitrate**).
Encryption *	Indicates whether the media stream being sent to and from the Conferencing Node towards the endpoint is encrypted.
Vendor *	Information about the endpoint's manufacturer and software.
Remote IP address *	The IP address of the system from which signaling from this endpoint is being sent and received. This may be the endpoint itself, or it may be a call control system if one is in use in your network.
Remote port *	The port on the system from which signaling from this endpoint is being sent and received.
Call ID *	A unique identifier that can be used to trace the call in the administrator log and support log. Select **View call logs** to see a filtered view of the support log showing only events containing this Call ID. Calls made via the Virtual Reception generate two separate participant calls but these both have the same Call ID. Calls made via the Pexip Distributed Gateway generate separate participant calls with different Call IDs.

* Only displayed when you have selected an individual participant to view.

Media streams

Media stream details are displayed when you view the details of an individual participant. (Note that media streams are not displayed for any Lync / Skype for Business participants that are connected to a Lync / Skype for Business meeting in a Fusion call.)

Type	Indicates whether the information is for the **Audio**, **Video**, or **Presentation** stream. When viewing the stream details of a completed call you may see multiple instances of each stream type (for example, if the participant had started presenting, stopped and then started presenting again). For in-progress calls you can only see a maximum of one instance of each stream type (reflecting what the participant is currently doing). Note that a presentation stream is not shown for Infinity Connect clients that are sending or receiving still images or PDF pages (as opposed to screen sharing, or receiving full motion presentation).
Start time	The date and time that the media stream started.
Node	The address of the Conferencing Node handling the media.
Tx codec	The format used by the Conferencing Node to encode the media stream being sent to the endpoint. ***Off*** indicates that no decodable media for this stream has been received in the last 10 seconds or so from the codec. For example, for an Infinity Connect client being used for presentation only, there will be no video transmitted. ***Off stage*** indicates that the participant is currently not being shown in the main video or thumbnails, so Pexip Infinity is not attempting to decode the media stream. If this field is blank, this may mean that Pexip Infinity has negotiated which codec to use but it is yet to receive any media from the endpoint to determine which codec it is actually sending.
Tx bitrate (kbps)	The quantity of data currently being sent from the Conferencing Node to the endpoint, in kilobits per second.
Tx resolution	The display resolution of the image being sent from the Conferencing Node to the endpoint.
Tx packets sent	The total quantity of packets sent from the Conferencing Node to the endpoint since the start of the conference.

Tx packets lost	The total quantity of packets sent from the Conferencing Node but not received by the endpoint. This value is reported to the Conferencing Node by the endpoint. Endpoints that do not support RTCP are not able to supply this information, so the value will always be **0**.
Tx jitter (ms)	The variation in the expected periodic arrival of packets being sent from the Conferencing Node to the endpoint, in milliseconds. This value is reported to the Conferencing Node by the endpoint. Endpoints that do not support RTCP are not able to supply this information, so the value will always be **0**.
Rx codec	The format used by the Conferencing Node decode the media stream being sent from the endpoint. ***Off*** indicates that no decodable media for this stream has been received in the last 10 seconds or so from the codec. For example, for an Infinity Connect client being used for presentation only, there will be no video transmitted. ***Off stage*** indicates that the participant is currently not being shown in the main video or thumbnails, so Pexip Infinity is not attempting to decode the media stream. If this field is blank, this may mean that Pexip Infinity has negotiated which codec to use but it is yet to receive any media from the endpoint to determine which codec it is actually sending.
Rx bitrate (kbps)	The quantity of data currently being received by the Conferencing Node from the endpoint, in kilobits per second.
Rx resolution	The display resolution of the image being received by the Conferencing Node from the endpoint.
Rx packets received	The total quantity of packets received by the Conferencing Node from the endpoint since the start of the conference.
Rx packets lost	The total quantity of packets sent from the endpoint but not received by the Conferencing Node.
Rx jitter (ms)	The variation in the expected periodic arrival of packets being received by the Conferencing Node from the endpoint, in milliseconds.

Viewing registrations

To see a list of all device aliases that are currently registered to the Pexip Infinity platform, go to **Status > Registrations**.

Devices can only register to Pexip Infinity if the alias it wants to register has been added to Pexip Infinity's list of allowed device aliases (**Service Configuration > Device Aliases**). For more information, see Registering devices to Pexip Infinity.

Registration details

The following information about each registration is available:

Field	Description
Alias	The alias that has been registered, including its URI scheme (e.g. sip:). Click on the alias to view detailed status information.
Username	The username associated with the alias that was used to authenticate the registration.
Node	The address of the node to which the device is registered.
Start time	The date and time the registration started.
Protocol	The protocol over which the device alias is registered, such as SIP, H.323 or WebRTC.
Is natted *	Indicates if the registered device is probably behind a NAT (its contact address is different from its source IP address).

* Only displayed when you have selected an individual alias to view.

Viewing historical information about conferences

To see a list of all the completed conferences on the Pexip Infinity platform, go to **Status > Conference History**. This shows a list of the most recent completed conferences (up to a limit of 10,000). To view details of a particular conference, click on the **Service name**.

To view information on conferences currently running on Pexip Infinity, see Viewing current conference status.

The following information is available for each completed conference:

Field	Description
Service name	The name of the Virtual Meeting Room, Virtual Auditorium, Virtual Reception, Test Call Service or Call Routing Rule. For Pexip Distributed Gateway calls, the rule name is followed by a unique identifier to distinguish between separate calls. For Virtual Receptions and Test Call Services, if there were multiple concurrent users of that service you will see a single instance of that service (rather than one instance per participant, as all participants are using the same service even though they cannot see or hear each other). Click on the service name to view more information.
Start time	The date and time that the first participant connected to the service.
End time	The date and time that the last participant's call ended.
Duration	The length of time that the conference or service was in use.
Participant count	The total number of participant calls made to this conference. Note that if a single participant disconnects from the conference and then reconnects to it, this will be counted as two participant calls.
Service type *	The type of conference, e.g. Virtual Meeting Room.
Service tag *	The unique identifier that an administrator has assigned to this service. If this field is blank, no tag has been assigned. For more information, see Tracking usage with a service tag.
Instant message count*	The total number of instant messages sent during the conference.

* Only displayed when you have selected an individual conference to view.

To view more information about the conference, click on the service name. You will then see 2 tabs for the selected conference: Participant history and Backplane history.

Participant history

The **Participant history** section lists all the participants that were in the conference.

For more details about a particular participant, including media stream statistics, click on the **Participant alias**. This takes you to the Participant history page.

Field	Description
Participant alias	The name of the user or the registered alias of the endpoint.
Start time	The date and time that the participant's call reached Pexip Infinity.
End time	The date and time that the participant's call ended.
Duration	The length of time that the participant was connected to Pexip Infinity. This includes time connected to the Virtual Meeting Room or Virtual Auditorium, and any time spent at the Virtual Reception or PIN entry screens.
Display name	The name that has been configured on the participant's endpoint.
System location	The system location of the Conferencing Node to which the endpoint is connected.

Field	Description
Role	**Host** indicates that either: • the conference had no PINs configured (in which case all participants had a role of **Host**) • the participant accessed the conference using the **Host PIN**. **Guest** indicates that the participant accessed the conference using the **Guest PIN**. **Unknown** indicates one of the following: • the participant reached the Virtual Reception but did not proceed to a Virtual Meeting Room or Virtual Auditorium • the participant reached the PIN entry screen but did not successfully enter a PIN • the call was via the Pexip Distributed Gateway. For more information, see About PINs, Hosts and Guests.

Backplane history

The **Backplane history** section provides information about the media streams being transmitted between Conferencing Nodes for the selected conference. Backplane links are unidirectional, so for a conference involving two Conferencing Nodes there will be two backplane links: one from node A to node B, and another from node B to node A.

Field	Description
Media node	The IP address of the Conferencing Node that is transmitting media. For details about the media streams being sent over a particular backplane link, click on the Media node's IP address.
Remote media node	The IP address of the Conferencing Node that is receiving media. Note that the remote media node of a merged Lync/SfB meeting is identified by the address of the Lync/SfB client that initiated the Lync/SfB meeting.
System location *	The system location of the Conferencing Node that is transmitting media.
Start time	The date and time that the connection was established.
End time	The date and time that the connection was brought down.
Backplane type	**Geographic** indicates that the two Conferencing Nodes are in different system locations. **Local** indicates that the two Conferencing Nodes are in the same system location.

* Only displayed when you have selected an individual media node to view.

Backplane media streams

Media stream details are displayed when you have selected an individual node to view.

Field	Description
Type	Indicates whether the information is for an **Audio**, **Video**, or **Presentation** stream.
Start time	The time that the media stream started.
End time	The time that the media stream ended.
Tx codec	The format used by the transmitting Conferencing Node to encode and decode the media stream being transmitted.
Tx bitrate (kbps)	The quantity of data currently being sent from the transmitting Conferencing Node to the recipient Conferencing Node for this particular media stream.
Tx resolution	The display resolution of the image being sent from the transmitting Conferencing Node.
Tx packets sent	The total quantity of packets sent from the transmitting Conferencing Node to the recipient Conferencing Node since the start of the conference.

Field	Description
Tx packets lost	The total quantity of packets sent from the transmitting Conferencing Node but not received by the recipient Conferencing Node.

Viewing historical information about participants

To see a list of all the calls made to the Pexip Infinity platform, go to **Status > Participant History**. This shows a list of all calls made to the 5,000 most recent completed conferences. To view details of a particular call, click on the **Participant alias**.

To view information on participants currently connected to Pexip Infinity, see Viewing participant status.

The following information is available:

Field	Description
Participant alias	The name of the user or the registered alias of the endpoint. Click on the participant alias to view detailed information about the call.
Service name	The name of the Virtual Meeting Room, Virtual Auditorium, Virtual Reception, Test Call Service or Call Routing Rule. For Pexip Distributed Gateway calls, the rule name is followed by a unique identifier to distinguish between separate calls.
Start time	The date and time that the participant's call reached Pexip Infinity.
End time	The date and time that the participant's call ended.
Duration	The length of time that the participant was connected to Pexip Infinity. This includes time connected to the Virtual Meeting Room or Virtual Auditorium and any time spent at the Virtual Reception or PIN entry screens.
Display name	The name that has been configured on the participant's endpoint.
Conference alias *	The alias that the participant dialed to access the service. If the participant included a PIN number in the dial string, this would also be included in the alias shown here.
System location	The system location of the Conferencing Node to which the endpoint is connected.
Signaling node *	The IP address of the Conferencing Node to which the endpoint is connected. This Conferencing Node will be handling the call signaling but may or may not be handling the media (for more information, see Handling of media and signaling in locally distributed conferences).
Media node *	The IP address of the Conferencing Node that is handling the call media for this endpoint (for more information, see Handling of media and signaling in locally distributed conferences).
Service type *	Indicates whether the participant was connected to a Virtual Meeting Room, Virtual Auditorium, Virtual Reception or the Pexip Distributed Gateway.
Role	**Host** indicates that either: • the conference had no PINs configured (in which case all participants had a role of **Host**) • the participant accessed the conference using the **Host PIN**. **Guest** indicates that the participant accessed the conference using the **Guest PIN**. **Unknown** indicates one of the following: • the participant reached the Virtual Reception but did not proceed to a Virtual Meeting Room or Virtual Auditorium • the participant reached the PIN entry screen but did not successfully enter a PIN • the call was via the Pexip Distributed Gateway. For more information, see About PINs, Hosts and Guests.
License count *	The number of licenses consumed by this participant. Media participants consume 1 license but API-only participants (e.g. Infinity Connect users who are not sending media) do not consume a license.

Field	Description
License type *	The type of license used, either "Port" (audio/video participant), "Audio" (audio-only participant — only applies if the system has audio licenses installed, otherwise port licenses are used for audio-only calls) or "None" (presentation or control only).
Streaming or recording device *	Indicates if the participant is a streaming or recording device.
Protocol *	The communication protocol being used by the endpoint.
Call direction *	**In**: the call was placed by an external endpoint and received by Pexip Infinity. **Out**: the call was placed by Pexip Infinity to an endpoint or other device.
Bandwidth (kbps) *	The maximum bandwidth, in kbps, negotiated for use between the Conferencing Node and the endpoint. Actual bandwidth used is shown in the Viewing historical information about participants section (**Tx bitrate** and **Rx bitrate**).
Encryption *	Indicates whether the media stream being sent to and from the Conferencing Node towards the endpoint is encrypted.
Vendor *	Information about the endpoint's manufacturer and software.
Remote IP address *	The IP address of the system from which signaling from this endpoint is being sent and received. This may be the endpoint itself, or it may be a call control system if one is in use in your network.
Remote port *	The port on the system from which signaling from this endpoint is being sent and received.
Call ID *	A unique identifier that can be used to trace the call in the administrator log and support log. Select **View call logs** to see a filtered view of the support log showing only events containing this Call ID. Calls made via the Virtual Reception generate two separate participant calls but these both have the same Call ID. Calls made via the Pexip Distributed Gateway generate separate participant calls with different Call IDs.
Disconnect reason *	The reason that the call was disconnected. This is provided by either the endpoint or Pexip Infinity, depending on how the call was terminated.

* Only displayed when you have selected an individual participant to view.

Media streams

Historical media stream details are displayed when you view the details of an individual participant.

Viewing usage statistics

Viewing graphs

To view the daily usage statistics per system location on:

- the total number of minutes used, and
- the highest number of concurrent ports used,

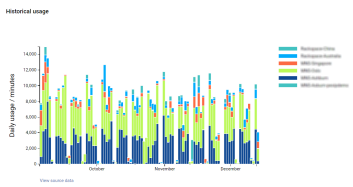
go to **Status > Usage Statistics**.

These graphs show statistics for up to the past 100 days, even if you have rebooted or upgraded your system during that time. Each day represents usage for the 24-hour period from midnight UTC.

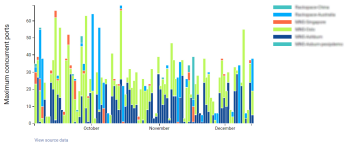
If there is no information available, for example if your system has been deployed in the last 24 hours, you will see the following message:

Information not yet available. This can take up to 1 hour.

Examples



Graph showing usage in minutes per day for a deployment spread across six locations



Graph showing the maximum number of concurrent ports used each day for a deployment spread across six locations

Viewing source data

To view the source data used to create each graph, click on the **View source data** link below the graph. This takes you to a page showing the source data in CSV format.

The source data goes back for the same amount of time as the administrator log, from which they are extracted. For more information, see About the administrator log.

Viewing LDAP sync template results

To view the status of ongoing or completed LDAP template synchronization processes, go to **Status > LDAP Sync Status**.

You are shown the details of the most recent or in-progress run of each LDAP synchronization template. Information shown includes:

- run status e.g. Initializing sync: connecting to LDAP, Syncing, Sync succeeded
- number of VMRs created, deleted, updated and unchanged
- number of devices created, deleted, updated and unchanged
- number of warnings
- last updated date and time.

If necessary, you can select a template to view the details of the last warning message it generated. Typical warnings include:

Warning	Comment
Error syncing with LDAP	This can occur if Pexip Infinity fails to connect to, or authenticate with, the LDAP server. In which case the Pexip Infinity support log will provide more information. Alternatively, this can be caused by invalid syntax in the template's **LDAP user filter** or **LDAP user search DN** fields.
Alias clash: alias held by an existing conference has been reassigned to the newly synced conference	This occurs when a template generates an alias that is identical to an existing alias that is already assigned to another VMR (either created manually or via a different template). In this case, the alias will be reassigned to the VMR currently being created/updated.
Duplicate alias: attempting to create the same alias more than once for the same conference	This occurs when the template attempts to create the same alias more than once for the same conference. Check the alias patterns for your template.

Warning	Comment
Conference clash: overwriting manually created conference	This occurs when a template generates a VMR name that already exists, and that VMR was created manually. In this case, the existing VMR with that name will be updated with the new settings generated from the template.
Sync template clash: overwriting conference created by another template	This occurs when a template generates a VMR name that already exists, and that VMR was created via a different template. In this case, the existing VMR with that name will be updated with the new settings generated from the template.
Validation error when creating conference	This occurs if the synchronization tries to set a field to an invalid value (e.g. a PIN containing letters rather than numbers, or an empty VMR name). The error message will include some addition parameters such as vmr_name, error and vmr_data which will help to identify the VMR, the field and source LDAP data that is causing the problem.

All of the warning messages generated from a sync run can be viewed in the Administrator log (**Status > Administrator Log**) as described below.

Viewing all historical sync results via the Administrator log

To view all historic details of completed VMR and device synchronization processes:

1. Go to the Administrator log (**Status > Administrator Log**).
2. Search for **Beginning VMR Sync** or **Completed VMR Sync** to see details of the start end and events corresponding to every VMR synchronization process that has occurred, or search for **Beginning Device Sync** or **Completed Device Sync** to see details of the start end and events corresponding to every device synchronization process.
   Alternatively you can search for **vmrsync** to see details of all log entries relating to all VMR and device synchronization processes.
   Note that you will see "Beginning" and "Completed" log entries for VMR and device syncing even if the template is not configured to perform both types of syncing. This is because both sync processes are always performed in case they need to delete VMRs or devices from previous synchronizations.

More information

For more information about bulk-provisioning VMRs and devices from an LDAP-accessible database, see:

- Provisioning VMRs and devices from Active Directory via LDAP
- Troubleshooting LDAP server connections

Viewing Conferencing Nodes

To see a list of all Conferencing Nodes currently configured on this Pexip Infinity, go to **Status > Conferencing Nodes**.

Each Conferencing Node has the following information available:

Field	Description
Name	The name of the Conferencing Node. Click on the name to view detailed status information.
IPv4 Address	The primary IP address of the Conferencing Node.
Secondary address	The optional secondary interface IPv4 address of the Conferencing Node.
System location	The physical location of the Conferencing Node.
Deployment status	**Deployment succeeded** indicates that the Conferencing Node has been automatically deployed. **Deployment failed** indicates that an automatic deployment has not been successful. **Manual** indicates that the ***Manual*** option was selected when the Conferencing Node was deployed (for more information see Deployment types. Even if the manual deployment was successful, this field will still say **Manual**; however you can see from the **Last contacted** and **Last updated** fields whether the Conferencing Node is now up and running and receiving configuration updates. Other statuses indicate that an automatic deployment is currently in progress.
Maintenance mode	Indicates whether the node is currently in maintenance mode.
Version	The version number of the Pexip Infinity software that is currently installed on this Conferencing Node.
Last contacted	The date and time that the Conferencing Node last contacted the Management Node. In general, Conferencing Nodes should contact the Management Node every minute or so. • **Green text** indicates that the contact was within the last 2 minutes. • **Orange text** indicates that the contact was between 2 and 5 minutes ago. • **Red text** indicates that the contact was more than 5 minutes ago.
Last updated	The last time that the Conferencing Node's configuration was updated.
Max HD connections/ Maximum capacity - HD connections*	An estimate of the total number of simultaneous high-definition (HD) 720p video calls this Conferencing Node can handle the media for (assuming it is not being used for any other types of call at that time.)†
Maximum capacity - audio connections*	An estimate of the total number of simultaneous audio-only calls this Conferencing Node can handle the media for (assuming it is not being used for any other types of call at that time.)†
Maximum capacity - SD connections*	An estimate of the total number of simultaneous standard-definition (SD) video calls this Conferencing Node can handle the media for (assuming it is not being used for any other types of call at that time.)†
Maximum capacity - Full HD connections*	An estimate of the total number of simultaneous full high-definition (Full HD) 1080p video calls this Conferencing Node can handle the media for (assuming it is not being used for any other types of call at that time.)†
Media load	An estimate of how much of its total capacity the Conferencing Node is currently using.† Note that when a Conferencing Node is in maintenance mode, it reports a media load of 100%. This is to indicate that there is no current capacity available.
Usage graph*	Shows media usage % statistics for this Conferencing Node for the last 4 weeks.

* Only displayed when you have selected an individual Conferencing Node to view.

† For more information on server capacity, see Capacity planning.

Viewing system location status

To see a list of all system locations currently configured, go to **Status > Locations**.

The following information about each location is available:

Field	Description
Name	The name of the location.
Max HD connections	The maximum number of HD connections that can be supported by all of the Conferencing Nodes assigned to this location.
Maximum capacity - audio connections *	The maximum number of audio-only connections that can be supported by all of the Conferencing Nodes assigned to this location.
Maximum capacity - SD connections *	The maximum number of SD connections that can be supported by all of the Conferencing Nodes assigned to this location.
Maximum capacity - Full HD connections *	The maximum number of Full HD connections that can be supported by all of the Conferencing Nodes assigned to this location.
Media load	An estimate of how much of the location's total capacity is currently being used.
Usage graph *	Shows media usage statistics for this location for the last 7 days. It indicates: • Maximum possible media load: this is shown as a light blue line. The maximum load is shown in terms of audio-only connections, which are the least resource intensive. (Other call types use relatively more resource — for example an HD call requires roughly 12 times the resources of an audio-only call. See Types of call for more information.) The maximum possible media load typically stays consistent over time, but can vary if Conferencing Nodes are added to, or removed from, the location. • Actual media load: this is shown as a dark blue line. This shows the resources that are actually being consumed over time, relative to the maximum possible media load.

* Only displayed when you have selected an individual location to view.

Note that when a location has no associated Conferencing Nodes (for example, after the location has just been created), the location reports a media load of 100%. This is to indicate that there is no current capacity available.

Viewing cloud bursting status

Go to **Status > Cloud Bursting** to see an overview of the media load of your primary locations (that contain your "always-on" Conferencing Nodes), and whether your overflow nodes and locations are in use.

- Any issues relating to your cloud bursting deployment will also be shown on this page.
- The list of primary locations only includes those system locations that are configured with a **Primary overflow location** that contains bursting nodes.
- An **approaching threshold** message is displayed in the **Available HD** connections column for the primary locations when the number of available HD connections is less than or equal to the bursting threshold plus two.

  This message changes to **bursting threshold reached** when the number of available HD connections is less than or equal to the bursting threshold (and therefore overflow nodes are started up).
- You can manually start any overflow nodes by selecting **Start** for the required node (the **Start** option is in the final column of the **Cloud overflow nodes** table).
- The status page dynamically updates every 15 seconds.

For more information see Dynamic bursting to the AWS cloud.

Viewing alarms

When there are active alarms on your Pexip Infinity deployment, a flashing red icon ▲ appears at the top right of each page of the Administrator interface. To view details of the alarms, click on this icon or go to the **Alarms** page (**Status > Alarms**).

Alarms remain in place for as long as the issue exists. After the issue has been resolved (for example, when an existing conference ends, therefore freeing up licenses) the associated alarm will automatically disappear from the **Alarms** page.

Multiple instances of the same type of alarm can be raised. For example if two Conferencing Nodes are not correctly synchronized to an NTP server, you will see an alarm for each node.

You can view alarms that have since been resolved by searching the Administrator log for **administrator.alarm**. This will show a list of all **Alarm raised** and **Alarm lowered** events.

An alarm is raised in each of the following situations:

Alarm	Level	Cause	Suggested resolutions
The Management Node does not have a TLS certificate	Critical	The Management Node has no associated TLS certificate.	Upload a TLS certificate and associate it with the Management Node.
A Conferencing Node does not have a TLS certificate	Critical	A Conferencing Node has no associated TLS certificate.	Upload a TLS certificate and associate it with the Conferencing Node. Alternatively, and if appropriate for your deployment, associate an existing certificate with your Conferencing Node. When doing this, the existing certificate should already contain a SAN (Subject Alternative Name) that matches your Conferencing Node's FQDN. See Managing a node's TLS server certificate for more information.
NTP not synchronized	Error	A node has failed to synchronize with the configured NTP servers.	Ensure that NTP is enabled on the Management Node, and that NTP servers are assigned to each location. See Enabling NTP on the Management Node for more information.
CPU instruction set not supported	Critical	A Conferencing Node has gone into maintenance mode because it was deployed on a server with an unsupported processor instruction set (e.g. SSE4.1).	Deploy the Conferencing Node on a server with AVX or AVX2.
License limit reached*	Warning	A Conferencing Node is unable to accept a call because there are not enough concurrent licenses available on the system at this time. For more information, see Pexip Infinity license installation and usage.	• Wait until one or more of the existing conferences have finished and the licenses have been returned to the pool. • Contact your Pexip authorized support representative to purchase more licenses.
Licenses expiring	Warning	One or more of your licenses is due to expire within the next 60 days.	Contact your Pexip authorized support representative to renew your licenses.

Alarm	Level	Cause	Suggested resolutions
Call capacity limit reached*	Warning	A call has not been accepted because all Conferencing Nodes that are able to take the media for this call are at capacity. Note: to understand how often this issue is occurring in your deployment, search the Administrator log for "insufficient capacity".	• Deploy more Conferencing Nodes in this location. • Move existing Conferencing Nodes onto more powerful servers. • Allocate more virtual CPUs for Conferencing Nodes on existing servers (if there are sufficient CPU cores). Note that the Conferencing Node will have to be rebooted for this to take effect. • Configure each location with a primary and secondary overflow location. For further information on capacity and how calls consume resources, see Hardware resource allocation rules.
Management Node limit reached	Warning	The Management Node does not have sufficient resources for the current deployment size (number of Conferencing Nodes).	Increase the amount of RAM and the number of virtual CPUs assigned to the Management Node. See the recommended hardware requirements in Server design recommendations.
Trusted CA certificates expiring	Warning	One or more of your trusted CA certificates is due to expire within the next 30 days.	Obtain and upload an updated certificate for the certificate authority.
TLS certificates expiring	Warning	One or more of your TLS certificates is due to expire within the next 30 days.	Obtain and upload an updated TLS certificate.
Incomplete TLS certificate chains	Warning	A TLS certificate has an incomplete chain of trust to the root CA certificate.	Obtain and upload the appropriate chain of intermediate CA certificates to the Management Node (the certificate provider normally provides the relevant bundle of intermediate CA certificates).
Syslog server inaccessible	Warning	A syslog server has been configured to use TCP or TLS but either is not responding to contact requests, or the connection has dropped.	• Check your network connectivity. • Check that the syslog server is running.
Configuration not synchronized	Warning	This alarm is raised if the Conferencing Node status “Last contacted” time has not been updated for 2 minutes.	Normally, Pexip Infinity synchronizes its configuration once per minute; if this fails to occur this may indicate network connectivity or routing issues between the Management Node and the Conferencing Node, which could be due to a malfunction or misconfiguration of devices such as routers or firewalls etc. Ensure that all Pexip nodes are fully routable to each other (full mesh) in both directions. See General network requirements.
Connectivity lost between nodes	Warning	Communication to a Pexip Infinity node has been lost. The alarm is raised after 10 seconds of lost connectivity. These alarms may be raised during an upgrade, and is expected behavior. They automatically clear as each node is upgraded to the new software version, has restarted and is ready to handle calls.	Check network connectivity and routing as for "Configuration not synchronized" above, or in the case of a software upgrade, wait for the upgrade process to complete.

Alarm	Level	Cause	Suggested resolutions
Hardware instability detected	Warning	Pexip Infinity has detected that the underlying VM infrastructure has paused the Pexip virtual machine. This is usually indicative of overcommitted hardware, which we do not support. Pexip Infinity, since it is a real time system, requires dedicated access to the underlying CPU and RAM resources of the hardware host.	Ensure that the Management Node and all Conferencing Node have dedicated access to their own RAM and CPU cores. See the recommended hardware requirements in Server design recommendations.

* When a license / resource subsequently becomes available (because a participant leaves a conference), the alarm is not cleared immediately; the alarm is cleared after the next participant successfully joins the conference.

Cloud bursting alarms

The following alarms may be raised in relation to issues with dynamic cloud bursting. See Dynamic bursting to the AWS cloud for more information about resolving these alarms.

Alarm	Level	Cause	Suggested resolutions
Cloud-bursting node found, but no corresponding Conferencing Node has been configured	Warning	This occurs when Pexip Infinity detects an AWS instance with a tag matching your system's hostname but there is no corresponding Conferencing Node configured within Pexip Infinity.	This message can occur temporarily in a normal scenario when deploying a new Conferencing Node and you have set up the VM instance in AWS but you have not yet deployed the Conferencing Node in Pexip Infinity. In this case, the issue will disappear as soon as you have deployed the Conferencing Node.
A location contains cloud bursting nodes, but no other locations are using it for overflow	Warning	A location contains some cloud overflow nodes, but no other locations are using it as an overflow location.	Set the location containing the cloud overflow nodes as the **Primary overflow location** of the locations containing your "always on" Conferencing Nodes.
Not authorized to perform this operation	Error	Pexip Infinity is not authorized to view instance data or to start and stop instances in AWS.	Ensure that an appropriate policy document is configured in AWS and is attached to the user that is being used by the Pexip platform.
Authentication failure while trying to communicate with Amazon Web Services	Error	Pexip Infinity cannot sign in to AWS.	Check that the **Access Key ID** and **Secret Access Key** that are configured in **Platform Configuration > Global Settings** match the **User Security Credentials** for the user you added within **Identity And Access Management** in the AWS dashboard.
Cloud bursting process encountered an unexpected error	Error	Pexip Infinity encountered an unexpected error while managing the cloud overflow nodes.	Check the status of your cloud bursting nodes within Pexip Infinity (**Status > Cloud Bursting**) and of your instances within AWS. Also check administrator and support log messages that are tagged with a log module Name of **administrator.alarm** to see additional error message information.

Viewing login history

To see a record of attempts to log in to the Pexip Infinity Administrator interface, go to **Status > Login History**.

The system displays the 100 most recent login attempts. It displays successful and failed attempts, and for each attempt shows the time of the attempt, the username that was submitted, and the address and port of the remote client.

Note that you are taken directly to the **Login History** page after logging in to the Pexip Infinity Administrator interface if the system has been configured with a login banner and certificate-based authentication has been enabled.

About the support log

The Management Node collates the logs from itself and all Conferencing Nodes and compiles these into the support log.

The support log records all events occurring across the Pexip Infinity deployment, including:

- configuration changes on the system
- conferences starting and ending
- participants joining and leaving conferences
- participants presenting in conferences
- SIP, H.323 and BFCP signaling
- device registration requests
- DNS lookups.

Sensitive information, such as the content of chat messages, is not logged.

Log timestamps always use UTC.

The information in the support log is retained during reboot and upgrade.

You can also configure Pexip Infinity to use an external syslog server. This allows you to use the syslog server's associated tools to collate and process the log entries to obtain specific information about real-time events, tailored to your requirements.

For more information on the content of the support log, see Log output.

Viewing the support log

To view the support log, go to **Status > Support Log**. If required, you can export the support log to a text file by selecting **Download**.

The log appears in the format:

```
syslog_time system originating_time level name details
```

where:

Field	Description
`syslog_time`	In the format: `year-month-dayThour:minute:second.millisecondUTC_offset` The time that the event was logged by syslog on the originating system.
`system`	The IP address or host name of the system that sent the log message.
`originating_time`	In the format: `year-month-day hour:minute:second,millisecond` The time at which the event occurred on the originating system.
`level`	The severity of the event. The levels, in order of increasing severity, are: • DEBUG • INFO • WARNING • ERROR
`name`	The system module producing the log output.
`details`	Information about the event, as a series of name=value pairs.

Searching the support log

You can filter the entire support log to only show messages that contain a particular string (for example, an IP address or alias) using the search box at the top left of the web page.

The search box also supports regular expressions (regex), so for example, you could search for:

- **^2015-10-07T09:[34][[:digit:]]:** to limit the displayed log messages to those emitted between 09:30:00 and 09:49:59 UTC on 7 October 2015
- **.*"Registration added".*"sip:alice@example.com"** to find all instances of "Registration added" messages for the SIP alias alice@example.com

Note that the results are not paginated.

File size

The support log has a maximum size of 2 GB, after which it will be overwritten, starting with the oldest entries. Up to 10 MB is available in a single page view on the Pexip Infinity Administrator interface; you can navigate through additional log pages using the buttons at the bottom left of the page. (Note that each page is a separate file, so the first page, which is the end of the file, may not be full.)

About the administrator log

The administrator log is a subset of the support log. It contains information about events occurring during the normal use of the system which may be of interest to a system administrator, notably:

- Conferencing Node deployment and communication status
- configuration changes on the system
- conferences starting and ending
- participants joining and leaving conferences
- participants presenting in conferences
- device registrations
- VMR imports
- system backups.

Sensitive information, such as the content of chat messages, is not logged.

Log timestamps always use UTC.

The information in the administrator log is retained during reboot and upgrade.

Viewing the administrator log

To view the administrator log, go to **Status > Administrator Log**. If required, you can export the administrator log to a text file by selecting **Download**.

The log appears in the format:

```
syslog_time system originating_time level name details
```

where:

Field	Description
`syslog_time`	In the format: `year-month-dayThour:minute:second.millisecondUTC_offset` The time that the event was logged by syslog on the originating system.
`system`	The IP address or host name of the system that sent the log message.
`originating_time`	In the format: `year-month-day hour:minute:second,millisecond` The time at which the event occurred on the originating system.
`level`	The severity of the event. The levels, in order of increasing severity, are: • DEBUG • INFO • WARNING • ERROR
`name`	The system module producing the log output.
`details`	Information about the event, as a series of name=value pairs.

Searching the administrator log

You can filter the entire administrator log to only show messages that contain a particular string (for example, an IP address, alias or service tag) using the search box at the top left of the web page.

The search box also supports regular expressions (regex), so for example, you could search for:

- **^2015-10-07T09:[34][[:digit:]]**: to limit the displayed log messages to those emitted between 09:30:00 and 09:49:59 UTC on 7 October 2015
- **.*"Registration added".*"sip:alice@example.com"** to find all instances of "Registration added" messages for the SIP alias alice@example.com

Note that the results are not paginated.

File size

The administrator log has a maximum size of 1 GB, after which it will be overwritten, starting with the oldest entries. Up to 10 MB is available in a single page view on the Pexip Infinity Administrator interface; you can navigate through additional log pages using the buttons at the bottom left of the page. (Note that each page is a separate file, so the first page, which is the end of the file, may not be full.)

Log output

The Pexip Infinity Management Node collates the logs from itself and all Conferencing Nodes and compiles these into the support log and the administrator log (which is a subset of the events in the support log).

To view these logs from the Pexip Infinity Administrator interface, go to **Status > Support Log** or **Status > Administrator Log**. If required, you can export the logs to a text file by selecting **Download**.

You can also use a syslog server to collate logs remotely.

Log timestamps always use UTC.

The log appears in the format:

```
syslog_time system originating_time level name details
```

where:

Field	Description
`syslog_time`	In the format: `year-month-dayThour:minute:second.millisecondUTC_offset` The time that the event was logged by syslog on the originating system.
`system`	The IP address or host name of the system that sent the log message.
`originating_time`	In the format: `year-month-day hour:minute:second,millisecond` The time at which the event occurred on the originating system.
`level`	The severity of the event. The levels, in order of increasing severity, are: • DEBUG • INFO • WARNING • ERROR
`name`	The system module producing the log output.
`details`	Information about the event, as a series of name=value pairs.

Administrator log system modules

The administrator log contains information about events occurring during the normal use of the system which may be of interest to a system administrator.

The following list shows typical events that are shown in the administrator log. In addition to these, logs may also contain one-off events, such as bulk configuration import and incident reports.

For the sake of brevity, the examples shown below feature only the `name` and `details` sections of the log message.

administrator.system

Logs when the Pexip Infinity application has started on this Conferencing Node. Also logs web-based activities such as:

- import of service configuration
- deployment of a Conferencing Node
- configuration synchronization towards Conferencing Nodes
- liveness status of Conferencing Nodes

Examples

```
Name="administrator.system" Message="Starting Infinity application." Pid="1650"
Name="administrator.system" Message="Deploying worker virtual machine." Address="192.168.0.2"
Name="administrator.system" Message="Successfully deployed worker virtual machine."
Address="192.168.0.2"
Name="administrator.system.configuration" Message="Conferencing node configuration updated."
Node="conferencingnode1"
Name="administrator.system.configuration" Message="Conferencing node appears unavailable."
Node="conferencingnode1" Last-Reported="2013-06-20 14:21:35"
Name="administrator.system.configuration" Message="Conferencing node successfully updated after being
marked unavailable." Node="conferencingnode1"
Name="administrator.system.connectivity" Message="Connectivity lost to node." Node="192.168.0.2" Last-
Reported="Mon Dec 9 15:28:12 2013"
Name="administrator.system.connectivity" Message="Connectivity re-established." Node="192.168.0.2" Last-
Reported="Mon Dec 9 15:28:12 2013"
```

administrator.configuration

This module logs when:

- configuration is added or changed
- a user logs in or logs out of the system, or fails to log in
- a participant is manually dialed into a conference
- license requests are processed

Messages

```
Configuration added
Configuration changed
Configuration deleted
User logged in
User logged out
User login failed
Initiated dial command
License activation request stored for offline processing
Processed license request
```

Parameters

Parameter	Definition
User	The user who made the change.
Type	The field that was changed/added.
ID	The value that was assigned.
Remote-Address	The IP address of the system from which the request was received.
Remote-Port	The port on the system from which the request was received.

Examples

```
Name="administrator.configuration" Message="Configuration added" User="admin" Type="Virtual Meeting
Room" ID="meet.alice"
Name="administrator.configuration" Message="User login failed" User="admins" Remote-Address="192.0.2.0"
Remote-Port="53991"
```

administrator.conference

This module logs conference activity.

Parameters

Each log entry includes information in the form of a series of name=value pairs. These are defined as follows:

Parameter	Definition
Call-id	An identifier that allows correlation of messages from the same participant URI but over different connections, for example covering both an API and a WebRTC connection.
Conference	The name of the service being connected to.
ConferenceAlias	The alias that was dialed in order to connect to the service.
Conversation-id	An identifier that allows correlation of messages across separate "calls" for video+audio, RDP, chat for Lync / Skype for Business connections.
Detail	The reason the call was disconnected (for disconnect events).
Direction	Either "in" or "out" as to whether this is an inbound or outbound call.
Duration	The duration of the call (for disconnect events).
Licenses	The number of licenses allocated for the call.
Location	The system location of the Conferencing Node that received the call.
Media-node	The IP address of the Conferencing Node handling the media processing for the call.
Participant	The name or alias of the conference participant.
Participant-id	Allows correlation of messages from the same participant URI per connection type, for example an API connection will have a different Participant-id from a WebRTC connection.
Protocol	The protocol of the call (may also indicate "BACKPLANE" when a backplane between Conferencing Nodes is established or removed).
Role	The participant's role (either "host" / "chair" or "guest").
Service-tag	The unique identifier for the service, if configured.
Service-type	This is one of the following: • `conference` for a Virtual Meeting Room • `lecture` for a Virtual Auditorium • `two_stage_dialing` for a Virtual Reception • `test_call` for a Test Call Service call • `gateway` for a Pexip Distributed Gateway call
Signaling-location	The name of the location handling the signaling.
Signaling-node	The IP address of the Conferencing Node handling the signaling for the call.
Vendor	The vendor string of the endpoint that placed the call.

Examples

Trigger for the log message	Example log entry
Logged when a participant tries to connect to a conference (at this stage it is the conference alias that is logged).	`Message="Participant attempting to join conference." ConferenceAlias="sip:meet.alice@example.com" Participant="sip:alice@example.com" Protocol="SIP" Direction="in" Participant-id="44be63a0-...c52"`

Trigger for the log message	Example log entry
A conference instance is created on a Conferencing Node. This occurs when the first participant attempts to join this conference on that Conferencing Node. The **Conference** parameter indicates the name of the service, not the alias being dialed.	`Message="Conference has been created." Conference="Meet Alice" Service-tag="" Service-type="conference"`
The participant joins the conference. This log message is issued before a role is assigned.	`Message="Participant has joined." Conference="Meet Alice" Service-tag="" Service-type="conference" ConferenceAlias="sip:meet.alice@example.com" Participant="sip:alice@example.com" DisplayName="Alice Smith" Protocol="SIP" Direction="in" Vendor="TANDBERG/257 (TE4.1.1.273710)" Call-id="fe7107ba-...16c" Conversation-id="fe7107ba-...16c" Participant-id="44be63a0-...c52" Location="London" Licenses="1" Signaling-node="192.168.0.1" Signaling-location="London" Media-node="192.168.0.2"`
Logged if conference creation fails.	`Message="Failed creating conference." Conference="meet" Service-tag=""`
Logged if the participant cannot join the conference. The **Reason** may vary as appropriate.	`Message="Participant failed to join conference." ConferenceAlias="sip:meet.invalid@example.com" Participant="sip:bob@example.com" Reason="Conference not found"`
A participant leaves the conference. Includes the **Duration** (in seconds) for which they were present in the conference. This message provides the definitive record of the length of time a call was connected to a conference.	`Message="Participant has disconnected." Conference="Meet Alice" Service-tag="" Service-type="conference" ConferenceAlias="sip:meet.alice@example.com" Participant="sip:alice@example.com" DisplayName="Alice Smith" Protocol="SIP" Direction="in" Call-id="fe7107ba-...16c" Conversation-id="fe7107ba-...16c" Participant-id="44be63a0-...c52" Location="London" Licenses="1" Signaling-node="192.168.0.1" Signaling-location="London" Media-node="192.168.0.1" Duration="63.123" Detail="Remote disconnect"`
A conference has stopped on a particular Conferencing Node. The **Duration** of this conference is given in seconds.	`Message="Conference has been stopped." Conference="Meet Alice" Service-tag="" Service-type="conference" Duration="62.234"`
A participant starts / stops presenting.	`Message="Participant is presenting." Conference="meet" Service-tag="" Participant="sip:chuck@example.com" Call-id="e5dbd44b...243"` `Message="Participant has stopped presenting." Conference="meet" Service-tag="" Participant="sip:chuck@example.com" Call-id="e5dbd44b...243"`

Trigger for the log message	Example log entry

These messages are logged when the Infinity Connect Mobile client or Infinity Connect client (or any third party using the Pexip Client API) controls participants in a conference.	`Message="Participant hold requested by API." Conference="meet" Service-tag="" Participant="sip:alice@pexip.com" Call-id="e5dbd44b...243" Conversation-id="e5dbd44b...243"`
	
	`Message="Participant resume requested by API." Conference="meet" Service-tag="" Participant="sip:alice@pexip.com" Call-id="e5dbd44b...243" Conversation-id="e5dbd44b...243"`
	
	`Message="Participant disconnect requested by API." Conference="meet" Service-tag="" Participant="sip:alice@pexip.com" Call-id="e5dbd44b...243" Conversation-id="e5dbd44b...243"`
	
	`Message="Participant mute/unmute requested by API." Conference="meet" Service-tag="" Participant="sip:alice@pexip.com" Call-id="e5dbd44b...243" Conversation-id="e5dbd44b...243"`
	
	`Message="Participant presentation receipt enabled by API." Conference="meet" Service-tag="" Participant="sip:alice@pexip.com" Call-id="e5dbd44b...243" Conversation-id="e5dbd44b...243"`
	
	`Message="Participant presentation receipt disabled by API." Conference="meet" Service-tag="" Participant="sip:alice@pexip.com" Call-id="e5dbd44b...243" Conversation-id="e5dbd44b...243"`
	
	`Message="Outgoing call requested" Conference="meet" Participant="sip:carol@pexip.com" Protocol="SIP" Role="chair"`
	
	`Message="Sending DTMF to participant." Conference="meet" Service-tag="" Participant="sip:alice@pexip.com" Call-id="e5dbd44b...243" Conversation-id="e5dbd44b...243" Digits="4"`

These messages are logged when participants attempt to join a PIN-protected conference.	`Message="PIN entry correct." Participant="sip:bob@example.com" ConferenceAlias="sip:meet@example.com" Call-id="27efdd36...6be" Role="host" Detail="Participant entered correct conference PIN and will be placed into the conference."`
	
	`Message="Participant entered incorrect conference PIN" RetriesRemaining="2" Participant="sip:bob@example.com" ConferenceAlias="sip:meet@example.com" Call-id="27efdd36...6be"`
	
	`Message="Participant exceeded PIN entry retries. Call will be disconnected." Participant="sip:bob@example.com" ConferenceAlias="sip:meet@example.com" Call-id="27efdd36...6be"`
	
	`Message="PIN entry timed out." Participant="sip:bob@example.com" ConferenceAlias="sip:meet@example.com" Call-id="a15dc445-...dfc" Detail="The correct PIN has not been received. Call will be disconnected."`

These messages occur when participants are involved in a PIN-protected conference.	`Message="Host participant joining." Participant="sip:alice@example.com" ConferenceAlias="sip:meet@example.com" Call-id="4d3ad9a1-...833" Role="chair" Detail="Participant will be automatically placed into the conference."`
	
	`Message="Conference host timeout." Participant="sip:bob@example.com" ConferenceAlias="sip:meet@example.com" Call-id="4d3ad9a1-...833" Detail="The conference host has not joined. Call will be disconnected."`
	
	`Message="Guest participant waiting." Participant="sip:bob@example.com" ConferenceAlias="sip:meet@example.com" Call-id="50a86594-...797" Role="guest" Detail="Waiting until participant is allowed to join the conference."`
	
	`Message="Guest participant joining." Participant="sip:bob@example.com" ConferenceAlias="sip:meet@example.com" Call-id="50a86594-...797" Role="guest" Detail="Participant will be automatically placed into the conference."`

These messages occur when automatically ending a conference	`Message="Last host left the conference and has not re-joined; conference will be terminated." Conference="Meet Alice" Service-tag=""`
	
	`Message="Removing participant due to conference termination" Participant="sip:bob@example.com" Conference="Meet Alice" Service-tag=""`

administrator.participantdialer

This module logs outbound calls placed from a conference or via the Pexip Distributed Gateway.

Examples

```
Message="Placing outbound call" ConferenceAlias="meet.alice@example.com" DisplayName="meet.alice"
Participant-alias="sip:bob@example.com" Protocol="SIP"
```

Logged when dialing out from a conference.

```
Message="Placing outbound call" Conference="Route to Lync:423a18c5-b06c-4654-9d34-e6b82f308b64"
ConferenceAlias="alice@example.com" DisplayName="Alice" Participant-alias="sip:carol@example.com"
Protocol="MSSIP"
```

Logged when making a call via the Pexip Distributed Gateway.

administrator.gatewayroutingrulematcher

This module logs the aliases that are matched against Call Routing Rules. The parsed or unparsed alias match type depends upon the rule's **Match against full alias URI** setting.

Example

```
Message="Alias matched gateway rule" Name="Route calls from Lync" Description="" Service-tag=""
Unparsed-alias="alice@example.com" Parsed-alias="alice@example.com" Matched-alias-type="Parsed-alias"
```

administrator.registration

This module logs endpoint registration activities.

Examples

```
Message="Registration added" Alias="sip:alice@example.com" Protocol="SIP" Registration-id="8695ea23-
66fb-41d2-9266-d16598f5d133" Natted="False"
```

Logged when a registration is added.

```
Message="Registration deleted" Alias="bob@example.com" Protocol="WebRTC" Registration-id="6936f540-6dbc-
40ac-9d55-5469bbf4e561" Reason="Registration expired"
```

Logged when a registration is removed.

administrator.vmrsync

This module logs VMR and device synchronization template activities.

Examples

```
Message="Beginning VMR Sync" template="All employees" creating_vmrs="219" deleting_vmrs="0" modifying_
vmrs="0" unchanged_vmrs="0"
```

Logged when a VMR template synchronization is initiated.

```
Message="Completed VMR Sync" template="Europe" status="sync_succeeded" vmrs_created="18" vmrs_
deleted="10" vmrs_updated="5" vmrs_unchanged="34" warnings="3" last_warning="Sync template clash:
attempt to create same conference multiple times"
```

Logged when a VMR template synchronization finishes.

administrator.apps.cloudbursting

This module logs cloud bursting activities.

Examples

```
Message="On-prem location 'Oslo' is overloaded! Starting instance i-0043088a9073b57e6 (stopped) in
location 'AWS eu-west-1'"
```

Logged when an overflow Conferencing Node is started up.

```
Message="Shutting down idle worker i-597394d2 (running) (uptime: 50 minutes)"
```

Logged when an overflow Conferencing Node is shut down.

Support Log system modules

Support logs provide a more detailed record of activity on the Pexip Infinity system.

support.sip:

Shows SIP signaling.

support.dns

Shows DNS lookups associated with SIP signaling.

support.bfcp

Shows BFCP signaling.

support.h323.q931, support.h323.ras, support.h323.h245

Shows H.323 signaling.

support.rest

Shows messages to and from the Pexip Client API (used by the Infinity Connect Mobile client, Infinity Connect client, and other third parties).

support.jinja2

Shows messages generated by the pex_debug_log filter when inserted into local policy jinja2 scripts.

Content of signaling messages

All log lines for signaling-related messages contain source and destination IP addresses and ports.

Multi-line messages are encoded on a single line with the character sequence `^M` representing a linebreak.

In addition to observing remote IP addresses, for H323 logging, a `Uuid` is present for H.225 (RAS) and H.245 signaling messages. These come from the same IP address but different ports, and the Uuid allows you to identify that they relate to the same call.

For BFCP signaling, all log lines are tied to a SIP `Call-id`.

Most of the content of the support log is the signaling messages themselves; however it also includes logging such as TCP and TLS connection attempts, successes, and failures.

support.participant

This module logs details about conference participants, including presentation stream activation, loss of incoming video, and call media statistics when a call is completed.

Example

An example of end-of-call media statistics is as follows:

```
Name="support.participant" Message="Media Stream destroyed" Participant="sip:alice@example.com" Call-
id="890b38520c27737c@192.168.0.1" Detail="Stream 0 (audio)
  RX: rate 54kbps loss 0.77% jitter 6ms
  TX: rate 61kbps loss 0.00% jitter 2ms
    codec MP4A-LATM rate 64kbps"
Name="support.participant" Message="Media Stream destroyed" Participant="sip:alice@example.com" Call-
id="890b38520c27737c@192.168.0.1" Detail="Stream 1 (video)
  RX: rate 1085kbps loss 0.00% jitter 3ms
  TX: rate 699kbps loss 0.00% jitter 5ms
    codec H264 rate 700kbps resolution 768x448 fps 30"
Name="support.participant" Message="Current participant is the presenter"
Participant="sip:bob@example.com" Call-id="c15b773a81af1740@192.168.0.1"
Name="support.participant" Message="Presentation started" Presenter="sip:bob@example.com"
Participant="sip:alice@example.com" Call-id="890b38520c27737c@192.168.0.1"
Name="support.participant" Message="Stopped sending presentation to participant" Last-
presenter="sip:bob@example.com" Participant="sip:alice@example.com" Call-
id=890b38520c27737c@192.168.0.1"
Name="support.participant" Message="Call is on hold" Participant="sip:alice@example.com" Call-
id="890b38520c27737c@192.168.0.1"
Name="support.participant" Message="Call has resumed" Participant="sip:alice@example.com" Call-
id="890b38520c27737c@192.168.0.1"
Name="support.participant" Message="Lost incoming video" Participant="sip:alice@example.com" Call-
id="890b38520c27737c@192.168.0.1"
```

Tracking usage with a service tag

Each Virtual Meeting Room, Virtual Auditorium, Virtual Reception, Call Routing Rule and registered device can be assigned a unique identifier, known as a service tag. You can then use this tag to track usage of the service, for example for billing purposes.

In most cases, if you are using service tags in your deployment you would use the Pexip Infinity Management API to obtain the information you require from the logs (for more information, see Log output). However, you can also search the Administrator log directly from the web-based Administrator interface (**Status > Administrator Log**) using the service tag to track usage of the service.

Example

A service provider is using the Pexip Infinity Management API to create Virtual Meeting Rooms automatically on behalf of customers. They assign each new VMR a user-friendly **Service name** - this is the name that will be presented to participants accessing the conference and will appear in Lync / Skype for Business users' contact lists. They also assign each VMR a randomly generated UUID as the **Service tag**. The service provider then uses the Service tag UUID to track usage of each VMR in order to bill each customer appropriately for using their services.

Pexip Infinity port usage and firewall guidance

The diagrams and tables below show the ports used when the Management Node and Conferencing Node connect to other devices.

Firewall, routing and NAT guidance

Note that in all Pexip Infinity deployment scenarios:

- All Pexip nodes must be fully routable to each other (full mesh) in both directions. This means that the Management Node must be able to reach every Conferencing Node, and each Conferencing Node must be able to reach every other Conferencing Node.
- Any internal firewalls must be configured to allow UDP port 500 and traffic using IP protocol 50 (ESP) in both directions between all Pexip nodes.
- There cannot be a NAT between any Pexip nodes.

When a secondary network address is configured on a Conferencing Node:

- The primary address is always used for inter-node communication to the Management Node and to other Conferencing Nodes.
- The secondary address is always used for signaling and media (to endpoints and other video devices).
- You can have a mixture of any number of single-interfaced and dual-interfaced Conferencing Nodes, providing all nodes can communicate with each other via their primary interfaces.

Management Node

Inbound

Protocol	Source-Port	Dest-Port	Description	Device
TCP	<any>	22	SSH *	SSH client
TCP	<any>	80	HTTP *	Web browser / API interface
TCP	<any>	443	HTTPS	Web browser / API interface
UDP	<any>	161	SNMP ‡	SNMP server
UDP	500	500	ISAKMP (IPsec)	Conferencing Node
ESP	n/a	n/a	IPsec / IP Protocol 50	Conferencing Node

Outbound

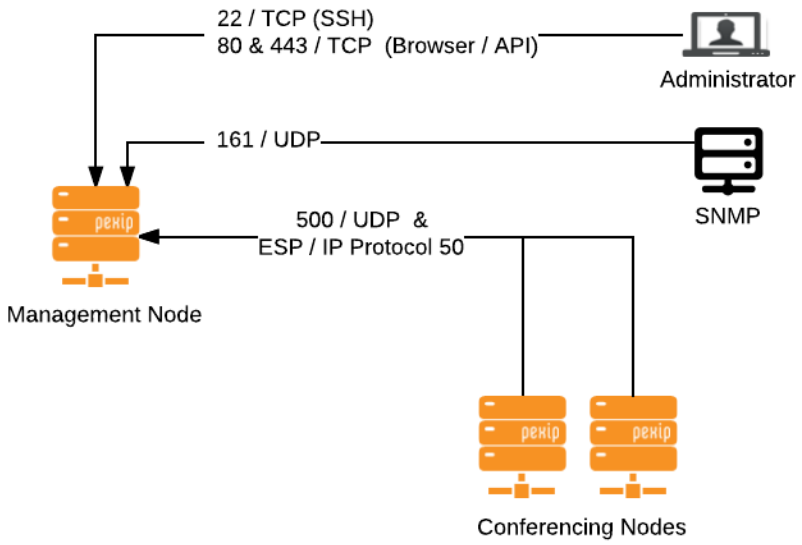
Protocol	Source-Port	Dest-Port	Description	Device
TCP/UDP	55000–65535	53	DNS	DNS server
TCP	55000–65535	389 / 636	LDAP ‡	LDAP server
TCP	55000–65535	443	HTTPS	vCenter Server and any ESXi host on which Conferencing Nodes may be deployed
TCP	55000–65535	443	HTTPS	Pexip Licensing server (pexip.flexnetoperations.com, 64.14.29.85)
TCP	55000–65535	443	HTTPS ‡	Incident reporting server (acr.pexip.com)
TCP	55000–65535	443	HTTPS ‡	Usage statistics (api.keen.io)
TCP	55000–65535	587	SMTP ‡	SMTP server
UDP	123, 55000–65535	123	NTP	NTP server
UDP	<any>	161 †	SNMP ‡	SNMP NMS
UDP	500	500	ISAKMP (IPsec)	Conferencing Node
UDP †	55000–65535	514 †	Syslog ‡	Syslog server
ESP	n/a	n/a	IPsec / IP Protocol 50	Conferencing Node

* Only required if you want to allow administrative access via this port.

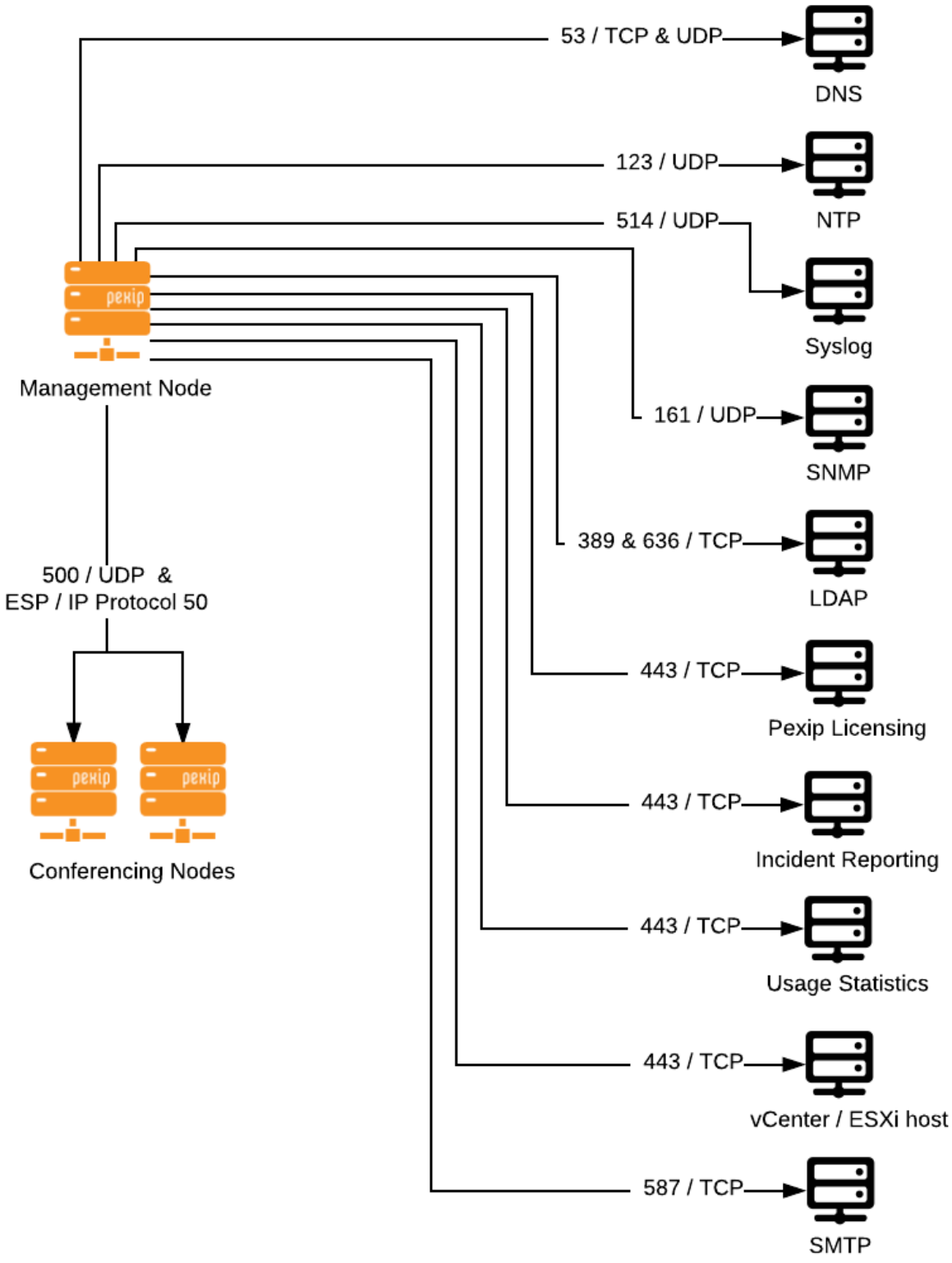
† Configurable by the administrator.

‡ Only applies if the relevant feature is configured.

Note also that the ephemeral port range (55000–65535) is subject to change.



Management Node inbound port usage



Management Node outbound port usage

Conferencing Nodes

Inbound

Protocol	Source-Port	Dest-Port	Description	Device
TCP	<any>	22	SSH *	SSH client
TCP	<any>	80	HTTP	Web browser / API interface / Lync / Skype for Business system (for conference avatar)
TCP	<any>	443	HTTPS	Web browser/ API interface / Infinity Connect Mobile client
TCP	<any>	1720	H.323 (H.225 signaling)	Endpoint / call control system
TCP	<any>	5060	SIP	Endpoint / call control system
UDP ‡	<any>	5060	SIP	Endpoint / call control system
TCP	<any>	5061	SIP/TLS	Endpoint / call control system
TCP	<any>	33000–39999 **	H.323 (Q.931/H.245 signaling)	Endpoint / call control system
TCP/UDP	<any>	40000–49999 **	RTP / RTCP / RDP / DTLS / RTMP / STUN / TURN	Endpoint / call control system / Lync / Skype for Business system / Infinity Connect ††
UDP	<any>	161	SNMP ‡	SNMP server
UDP	500	500	ISAKMP (IPsec)	Management Node / Conferencing Node
UDP	<any>	1719	H.323 (RAS signaling)	Endpoint / call control system
ESP	n/a	n/a	IPsec / IP Protocol 50	Management Node / Conferencing Node

Outbound

Protocol	Source-Port	Dest-Port	Description	Device
TCP/UDP	55000–65535	53	DNS	DNS server
TCP	55000–65535	443	HTTPS ‡	Incident reporting server (acr.pexip.com)
TCP	33000–39999 **	1720	H.323 (H.225 signaling)	Endpoint / call control system
TCP/UDP	33000–39999 **	5060	SIP	Endpoint / call control system
TCP	33000–39999 **	5061	SIP/TLS	Endpoint / call control system
TCP	33000–39999 **	<any>	H.323 (Q.931/H.245 signaling)	Endpoint / call control system
TCP/UDP	40000–49999 **	<any>	RTP / RTCP / RDP / DTLS / RTMP / STUN / TURN	Endpoint / call control system / Lync / Skype for Business system / Infinity Connect ††
TCP	40000–49999 **	1935	RTMP	RTMP streaming server
TCP (TLS)	55000–65535	443 / 8057 ‡‡	PSOM (PowerPoint presentation from Lync/SfB)	Lync/SfB Web Conferencing service
TCP (TLS)	55000–65535	443	HTTPS (PowerPoint presentation from Lync/SfB)	Lync/SfB Front End server or Edge server
UDP	123, 55000–65535	123	NTP	NTP server
UDP	<any>	161 †	SNMP ‡	SNMP NMS

Protocol	Source-Port	Dest-Port	Description	Device
UDP	500	500	ISAKMP (IPsec)	Management Node / Conferencing Node
UDP †	55000–65535	514 †	Syslog ‡	Syslog server
UDP	33000–39999 **	1719	H.323 (RAS signaling)	Endpoint / Call control system
UDP	40000–49999 **	3478 †	STUN / TURN	STUN / TURN server
ESP	n/a	n/a	IPsec / IP Protocol 50	Management Node / Conferencing Node

* Only required if you want to allow administrative access via this port.

† Configurable by the administrator.

** Configurable via the Media port range start/end and Signaling port range start/end options (see About global settings).

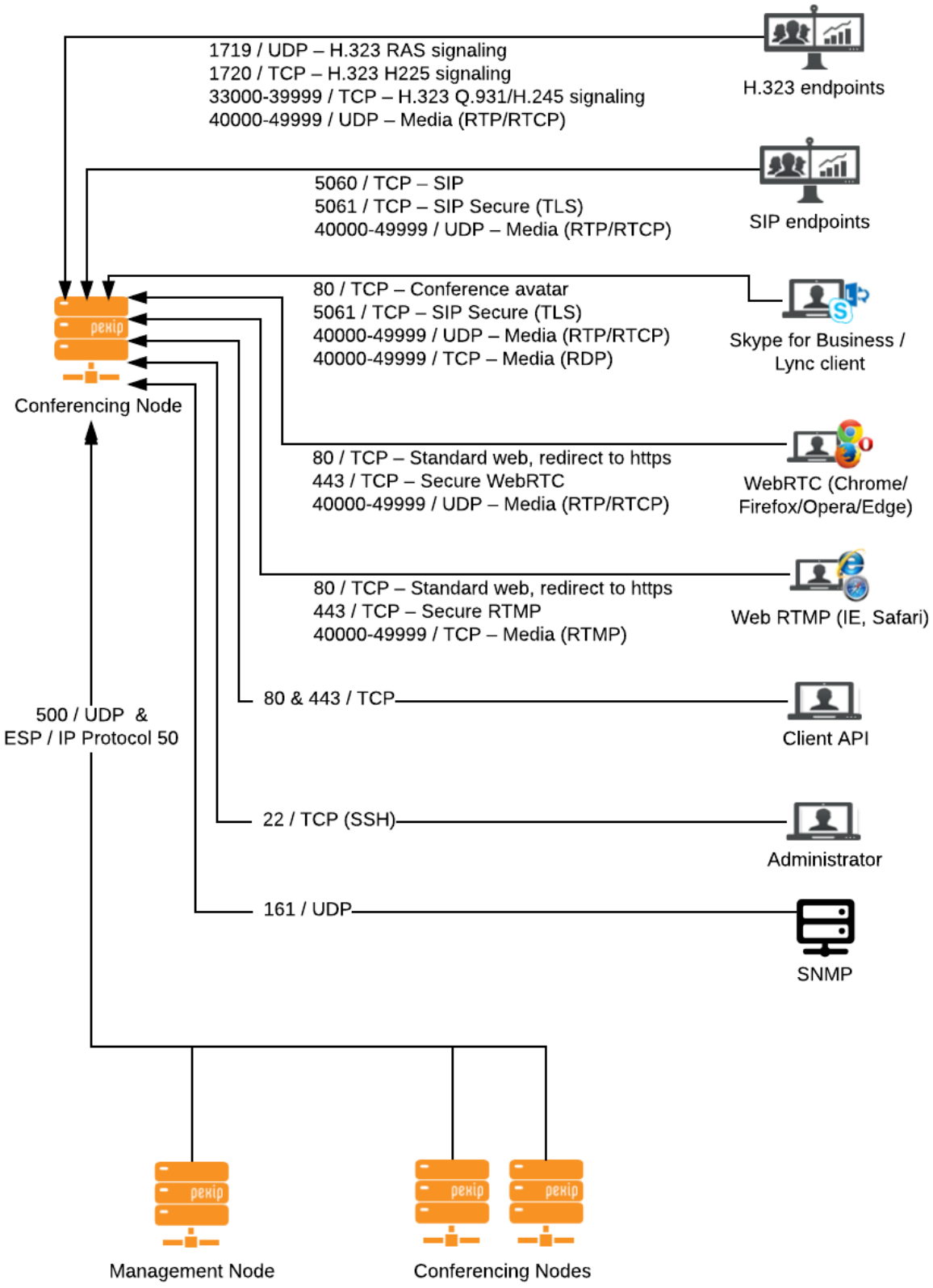
†† Infinity Connect web, mobile and desktop (installable) clients

‡ Only applies if the relevant feature is configured.

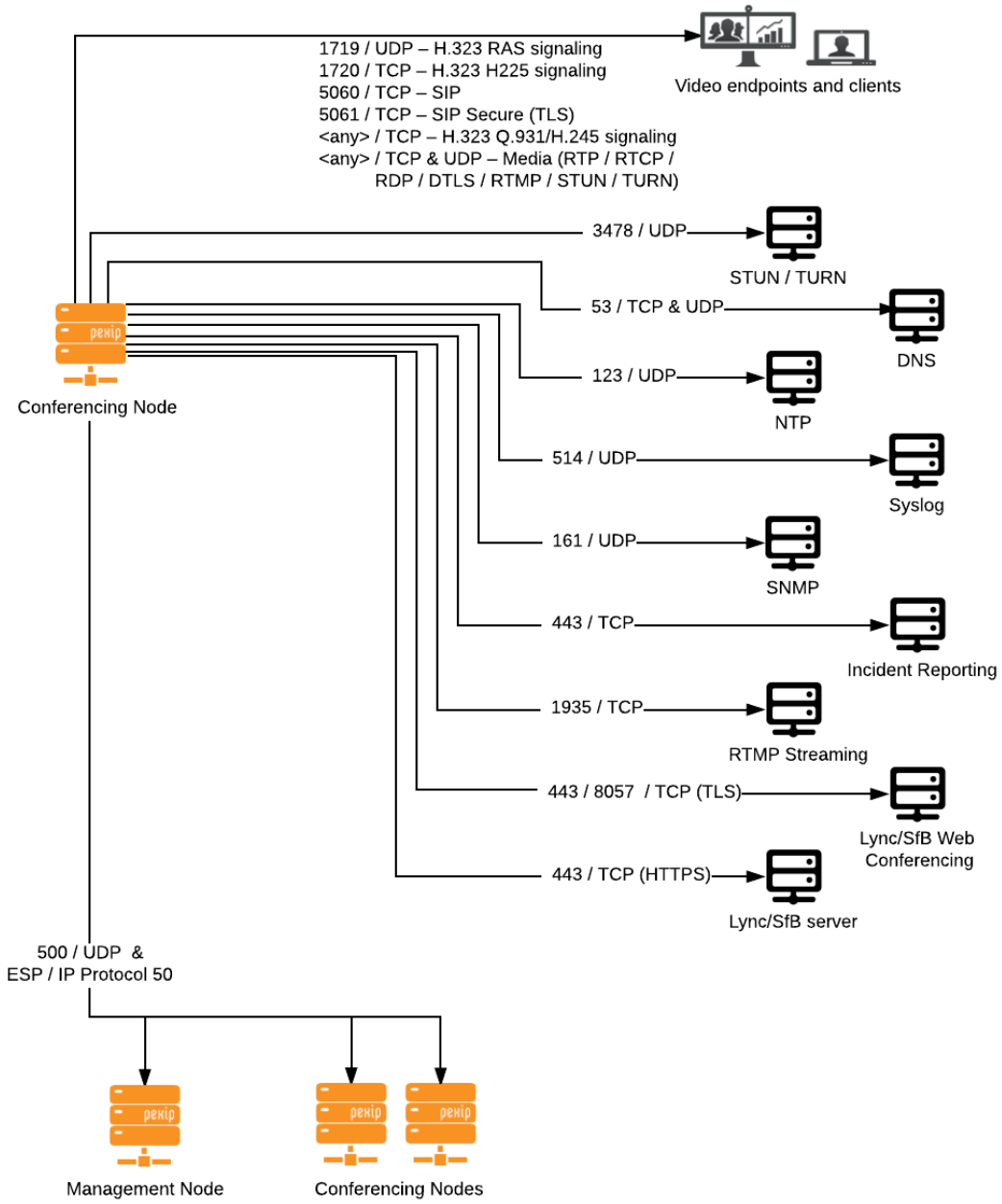
‡‡ Typically 443 for Web Conferencing Edge and 8057 for a Lync/SfB Front End server / FEP.

Note also that:

- ICE calls allocate 4 ports per media line/stream.
- The ephemeral port range (55000–65535) is subject to change.



Conferencing Node inbound port usage



Conferencing Node outbound port usage

Troubleshooting the Pexip Infinity platform

This section contains a list of symptoms, possible causes and suggested resolutions for some issues you may experience when using the Pexip Infinity platform.

It includes the following sections:

- Pexip Infinity deployment and upgrading
- Joining a conference and viewing content
- Conference connectivity and TLS issues
- Pexip Infinity administration
- Infinity Connect clients

Help in resolving LDAP-specific issues is contained in a separate topic: Troubleshooting LDAP server connections.

Help in resolving Lync / Skype for Business issues is contained in a separate topic: **Troubleshooting and limitations with Lync / Skype for Business and Pexip Infinity integrations**.

For a list of devices that are supported by Pexip Infinity, including any known issues, see Interoperability.

Pexip Infinity deployment and upgrading

Symptom	Possible cause	Resolution
During upgrade, one or more Conferencing Nodes are stuck with a status of "waiting for calls to clear", but there are no active calls reported on the Management Node.	In some circumstances a call will not clear properly from the Conferencing Node.	Reboot the Conferencing Node.
During automatic deployment of a Conferencing Node using VMware, the following message is given: "Server raised fault: 'Current license or ESXi version prohibits execution of the requested operation.'"	You are using a version of VMware that does not support use of the VMware API, which is required for automatic deployment of Conferencing Nodes.	Either: • Upgrade your VMware deployment to one that supports the VMware API. For more information, see VMware editions. • Deploy the Conferencing Node manually. For more information, see Manually deploying a Conferencing Node on an ESXi host.
An automatic deployment of a Conferencing Node using vCenter as the VM manager fails at 22%.	During automatic deployment, the Management Node uploads the VM image directly to the ESXi host using HTTPS on port 443/TCP. If this port is not accessible, the upload will fail.	Ensure that all ESXi hosts on which Conferencing Nodes are being deployed are accessible from the Management Node using HTTPS on port 443/TCP.

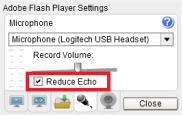
Symptom	Possible cause	Resolution
A Conferencing Node does not accept calls even though it is powered on and is contactable on the network.	If time is not properly synchronized between the Management Node and the host server, certificates issued by the Management Node may be invalidated by Conferencing Nodes within the same Pexip Infinity deployment. As a result, the Conferencing Nodes will not communicate properly with the Management Node, causing calls to fail.	Ensure all virtual machines (VMs) (i.e. the Management Node and all Conferencing Nodes) within the Pexip Infinity platform, and the host servers on which they are running, are using accurate times according to the public or private standard NTP clock. We strongly recommend that you configure at least 3 distinct NTP servers or NTP server pools in each instance to ensure proper synchronization. To synchronize time on Pexip Infinity: 1. Synchronize time on host servers (how to do this will vary depending on the hypervisor; for instructions, see the relevant hypervisor installation guide). 2. Enable NTP on Management Node. 3. Reboot all VMs.
A newly deployed Conferencing Node does not accept calls and its last contacted status on the Management Node shows "Never", even though it is powered on and is contactable on the network.	After deploying a new Conferencing Node, it takes approximately 5 minutes before the node is available for conference hosting and for its status to be updated on the Management Node. (Until it is available, the Management Node will report the status of the Conferencing Node as having a last contacted and last updated date of "Never".)	Wait for the Conferencing Node to finish initializing.
I entered the wrong information while running the installation wizard.		On server-based deployments you can re-run the installation wizard by following the instructions in Re-running the installation wizard. Do not re-run the installation wizard on cloud-based deployments (AWS or Azure) in order to change Management Node configuration data such as its IP address or hostname. To change such data you must terminate the existing instance and deploy a new Management Node instance. You should only re-run the installation wizard on cloud-based deployments if you need to reset the web administration password (and then you should not change any of the other configuration data).
A new Management Node or Conferencing Node does not work. It was created by cloning it through VMware.	You cannot use cloning to create Management Nodes and Conferencing Nodes.	Create all Management Nodes according to our instructions (see Installation overview for links to the appropriate instructions for your hypervisor). Create all Conferencing Nodes by following the instructions in Deploying new Conferencing Nodes.

Symptom	Possible cause	Resolution
A newly-deployed Conferencing Node has gone into maintenance mode with the message "CPU instruction set is not supported; system will be placed in maintenance mode".	The Conferencing Node has been installed on a system that does not meet the CPU instruction set requirements.	Do one of the following: • Migrate the Conferencing Node to another server. • Re-install the Conferencing Node on another server. • If the instruction set is limited because you are using EVC mode (or equivalent), increase the minimum EVC level to L4 (Sandy Bridge).

Joining a conference and viewing content

Symptom	Possible cause	Resolution
Participants cannot join a conference due to insufficient capacity. • When users attempt to join a conference they get a message saying "Participants cannot join a conference due to insufficient capacity." • There is an alarm "Call capacity limit reached". • The admin log is reporting "Participant failed to join conference" and "resource unavailable".	All Conferencing Nodes that are able to take the media for this call are at capacity.	• Deploy more Conferencing Nodes in this location. • Move existing Conferencing Nodes onto more powerful servers. • Allocate more virtual CPUs for Conferencing Nodes on existing servers (if there are sufficient CPU cores). Note that the Conferencing Node will have to be rebooted for this to take effect. • Configure each location with a primary and secondary overflow location. For further information on capacity and how calls consume resources, see Hardware resource allocation rules.
	The participant has dialed in to the Conferencing Node while it is still starting up and an internal capacity-checking tool is running.	The Administrator should place the Conferencing Node into maintenance mode if it is expected to be powered off and on. If the Conferencing Node is not in maintenance mode while the capacity-checking tool is running the participant should wait a few moments and then attempt to join the conference.
	The Virtual Meeting Room or Virtual Auditorium has a participant limit applied, and this limit has been reached.	Increase the participant limit, if appropriate.
Participants cannot join a conference due to an invalid license. • When users attempt to join a conference they get a message saying "Participants cannot join a conference due to an invalid license." • The admin log is reporting "Participant failed to join conference" and "no valid license available".	If your Pexip Infinity reports an **invalid license**, this could mean that: • the license has not been activated • the existing license has expired • the existing license has become corrupt (this could occur, for example, if the Management Node reboots after an upgrade and comes back up on a different physical blade with a new MAC address).	• Check the status of your licenses from the **Licensing** page (**Platform Configuration > Licenses**). • Contact your Pexip authorized support representative for assistance. For more information, see Pexip Infinity license installation and usage.

Symptom	Possible cause	Resolution
Participants cannot join a conference due to insufficient licenses. • When users attempt to join a conference they get a message saying "Participants cannot join a conference due to insufficient licenses." • There is an alarm saying "License limit reached". • The admin log is reporting "Participant failed to join conference" and "license limit reached".	There are not enough port licenses available on the system at this time. For more information, see Pexip Infinity license installation and usage.	• Wait until one or more of the existing conferences have finished and the licenses have been returned to the pool. • Contact your Pexip authorized support representative to purchase more licenses.
An H.323 endpoint has its bandwidth restricted when joining a conference via a Virtual Reception, or when placing a call using the Distributed Gateway after first connecting to a Virtual Reception.	If there has been a bandwidth restriction placed on the Virtual Reception, any H.323 endpoints using that service will not be able to subsequently increase their bandwidth, even after being transferred to a Virtual Meeting Room or using a Call Routing Rule that has a higher (or no) limit.	• Make the call using a SIP endpoint or Infinity Connect client. • Do not place a bandwidth restriction on the Virtual Reception.
Presentations do not display full screen	If the presentation being shared is either: • an application that is not in full-screen mode • a full screen image that is being sent from a non-standard aspect ratio screen then the image being sent may have a non-standard aspect ratio. In order to send the image inside a standard resolution window (for example 640x480 [4:3]) or 1280x720 [16:9]), the endpoint may add horizontal or vertical mattes (also known as letterboxing or pillarboxing respectively).	Ensure that presenters always either: • share their entire screen, or • share individual applications when they are in full-screen mode only.
Images are not displaying as expected: • they are being cropped • they have black bars at the top or sides	Endpoints send and display video images and presentations in various aspect ratios, most commonly 16:9 and 4:3. If there is a difference between the aspect ratios of the sending and receiving endpoints, then the endpoint and/or Pexip Infinity may crop the image or add vertical or horizontal mattes.	For more information, see Changing aspect ratios.
Main video on a Cisco E20 freezes when content is received	The following issue may occur when additional video codecs (e.g. H.264 High profile) have been enabled on the Pexip Infinity deployment: When an E20 has been dialed out to from a VMR, and a participant in the VMR presents content, the main video on the E20 freezes and no presentation is shown. The E20 will still send video as normal, and audio will flow in both directions. This occurs because more codecs are offered to the E20 than it is able to cope with.	

Symptom	Possible cause	Resolution
Participants keep hearing themselves repeated back after a short delay. This happens in a conference with one or more other participants connected using Infinity Connect via Internet Explorer, Edge, Safari or Firefox, and who are using their computer's microphone and speakers.	Internet Explorer, Edge, Safari and Firefox do not have adequate echo cancellation, and in certain circumstances may experience a delay in playing audio. When this happens, sounds played through the computer's speakers are picked up by the computer's microphone and replayed back to other participants.	Participants using Internet Explorer, Edge, Safari or Firefox should: • use a headset • mute themselves when not speaking • consider using Chrome or the Infinity Connect desktop client instead. Participants using Flash-based browsers should also ensure echo cancellation is enabled: 
In-band DTMF tones may not be detected if they are input too quickly.	False detections can be caused, for example, by poor line quality, line noise and echo.	This is best resolved through using out-of-band DTMF tones.
Participants are disconnected from conferences and "Backplane disconnected" messages are recorded in the administrator log.	VMware snapshots were being taken or deleted while conferences were in progress. Taking or removing snapshots can cause virtual machines to become unresponsive. For more information, see this VMware knowledge base article.	Only create and delete VMware snapshots at a time of minimal usage.

Conference connectivity and TLS issues

Symptom	Possible cause	Resolution
Calls fail when dialing out to, or receiving calls from, video network infrastructure devices, such as a Cisco VCS or CUCM. Calls may fail immediately, or after a period of time, and SSL alerts are raised in the support log with an "unsupported certificate" description.	Either the external system or the Conferencing Node is verifying the other party's certificate and it is rejecting the connection because the certificate does not have **client** authentication properties.	Ensure that the certificates on your external systems and on your Conferencing Nodes contain "TLS Web Client Authentication" Enhanced Key Usage properties (see Mutual TLS authentication and client/server certificates).
A Cisco MXP intermittently puts a call on hold immediately after resuming it.	This is due to a bug in the MXP where a race condition exists between the resume message and the session refresh re-INVITE message.	Set the session refresh configuration on your call control system to a value that avoids this race occurring.
After upgrading to version 10 or later, some Cisco MXPs may no longer be able to connect over TLS to Pexip Infinity.	This can occur if the endpoint connects directly to a Conferencing Node (e.g. because it is registered to Pexip Infinity, or it is using IP dialing).	Add to your Conferencing Node server certificates some custom DH parameters that define a DH group of size <=2048 bits. See Managing a node's TLS server certificate for information about how to do this.

Pexip Infinity administration

Symptom	Possible cause	Resolution
Log timestamps appear to be inaccurate or log entries appear to be out of sequence.	Time is not properly synchronized between the Management Node, Conferencing Nodes and their host servers, causing different systems to use different timestamps. This could be because: • Insufficient NTP servers or NTP pools have been configured on a host server or the Management Node (we recommend a minimum of 3). • One or more NTP servers are unreachable or have inaccurate time themselves. • NTP is otherwise not configured according to our recommendations.	Ensure all virtual machines (VMs) (i.e. the Management Node and all Conferencing Nodes) within the Pexip Infinity platform, and the host servers on which they are running, are using accurate times according to the public or private standard NTP clock. We strongly recommend that you configure at least 3 distinct NTP servers or NTP server pools in each instance to ensure proper synchronization. To synchronize time on Pexip Infinity: 1. Synchronize time on host servers (how to do this will vary depending on the hypervisor; for instructions, see the relevant hypervisor installation guide). 2. Enable NTP on Management Node. 3. Reboot all VMs.
How do I register Pexip Infinity to a gatekeeper?		You don't. Instead, you configure your call control system to route calls to Pexip Infinity. See Call control.
Oracle Acme Packet SBC has the error "Message Too Large".		Increase the following configuration parameters on the Acme Packet SBC: • `sip-message-len 16000` • `option +max-udp-length=0`
VMware datastore is showing disk I/O alarms.		Enabling SIOC on your datastores might help. For more information, see this VMware knowledge base article.

Infinity Connect clients

Symptom	Possible cause	Resolution
Participants are unable to use Infinity Connect clients. Infinity Connect Web App and Infinity Connect desktop client users are presented with a message "Server error! Please try again".	Support for these applications has been disabled.	Enable support as follows: 1. Go to **Platform Configuration > Global Settings**. 2. Select the **Enable support for Pexip Infinity Connect and Mobile App** checkbox.
Participants attempting to connect to a conference using the Infinity Connect Mobile client for iOS get a message "Certificate Error. Please contact your system administrator."	Your deployment does not use valid, trusted certificates. Infinity Connect Mobile clients require deployments with HTTPS and valid certificates.	Install valid certificates. For more information, see Managing TLS and trusted CA certificates.

Symptom	Possible cause	Resolution
Participants attempting to connect to a conference using the Infinity Connect Mobile client for Android get a message "Unable to connect to the server".	The user has entered an invalid address.	Check the address.
	Your deployment does not use valid, trusted certificates. Infinity Connect Mobile clients require deployments with HTTPS and valid certificates.	Install valid certificates. For more information, see Managing TLS and trusted CA certificates.
	The user is attempting to connect to a reverse proxy that requires authentication. They are using Version 3.0.11 or earlier of the Android client, which cannot provide authentication credentials.	Users should either: • Upgrade to version 3.0.12 or later. • Connect directly to a Conferencing Node.
Participants using an Infinity Connect client cannot see the chat window.	Chat has been globally disabled.	Enable chat.

Guest participants using an Infinity Connect client cannot see the chat window, participant list, or presentation.	Guest participants are not allowed in to the conference if the conference is locked, or until the first Host participant has joined using video or audio, or has joined as control-only and has explicitly started the conference.	• Allow an individual guest to join the conference.
		• Unlock the conference, allowing all guests to join.
		• Connect at least one endpoint to the conference as a Host participant either as:
		  ○ a video or audio participant, or
		  ○ a control-only Infinity Connect participant and then elect to start the conference manually.

Users cannot access the Infinity Connect Web App home page when using Internet Explorer.	Internet Explorer may be accessing the site in Compatibility View.	Ensure that the target domain is not in Internet Explorer's list of websites added to Compatibility View.
Users of the Infinity Connect Web App via Edge, Internet Explorer or Safari get no main video when connecting to a conference via a reverse proxy. Other conference participants see a muted camera icon in place of the participant's video.	To send and receive audio/video media, Internet Explorer, Edge, and Safari browsers need a direct TCP connection to a Conferencing Node (as they use the RTMP protocol and thus cannot use ICE/TURN). For more information, see **Using Infinity Connect from outside your network**.	• Connect over VPN. • Use a different browser.

Troubleshooting LDAP server connections

Pexip Infinity can be configured to connect to a Windows Active Directory LDAP server, or any other LDAP-accessible database, in order to:

- bulk-provision individual Virtual Meeting Rooms or devices for every member of the directory
- authenticate and authorize the login accounts that are allowed to connect to the Pexip Infinity Administrator interface or the Pexip Infinity API.

This section explains how Pexip Infinity connects to the LDAP server, and provides guidance on how to troubleshoot connection problems.

Connecting to the LDAP server

When resolving the **LDAP server address**, the system supports DNS SRV and DNS A/AAAA lookups. The system always tries in the first instance to set up a TLS connection with the LDAP server. If that fails it may fall back to a TCP connection if allowed.

To establish a TLS connection, the Pexip Infinity platform must trust the certificate presented by the LDAP server i.e. the LDAP server's certificate must be signed by an authority within the Pexip Infinity trusted CA certificates store. In addition, the resolved LDAP server address must match the CN (common name) contained within the certificate presented by the LDAP server. Note that the Pexip Infinity platform does not support mutual authentication — it will not supply its server certificate to the LDAP server.

The system will connect to the port returned by an SRV lookup, otherwise it will connect to 389 (TCP) or 636 (TLS).

Connection process

If the **LDAP server address** is configured as an IP address, the system will connect directly to the given address, otherwise it treats it as a domain or FQDN and attempts to resolve the address via DNS lookups in the following sequence:

1. Perform a DNS SRV lookup against _ldaps._tcp.<**LDAP server address**>.
2. Perform a DNS SRV lookup against _ldap._tcp.<**LDAP server address**>.
3. Perform a DNS A/AAAA lookup against <**LDAP server address**>.

When a DNS lookup is successful, the system will first attempt to establish a TLS connection with the server at the returned address. If the TLS connection attempt fails, the system will then attempt a TCP connection, but only if **Allow insecure transport** is enabled. Only TLS connections are attempted as a result of _ldaps lookups.

If multiple addresses are returned by SRV lookups, the system will attempt to connect to each address in priority order.

Connectivity error messages and using the support log

Diagnostic information is also recorded in the support log (**Status > Support Log**).

When Pexip Infinity connects successfully to the LDAP server, the support log will contain an entry similar to this:

2015-06-05T11:15:00.550+00:00 mgmt 2015-06-05 11:15:00,550 Level="INFO" Name="support.ldap" Message="Successfully connected to LDAP server" Address="server.example.com" Uri="ldaps://server.example.com"

Unable to contact the LDAP server

If Pexip Infinity cannot contact the configured LDAP server, the support log will contain an entry similar to this:

2015-06-05T08:40:29.707+00:00 mgmt 2015-06-05 08:40:29,704 Level="INFO" Name="support.ldap" Message="Failed connecting to LDAP server" Address="server.example.com" Reasons="
ldaps://server.example.com : Can't contact LDAP server
ldap://server.example.com : Can't contact LDAP server"

Ensure that the server is available at the configured address and, if the server address is specified by domain name or FQDN, ensure that DNS records exist and resolve to the correct address.

Connection errors: TLS certificate issues

If Pexip Infinity can reach the configured LDAP server, but cannot connect to it due to TLS certificate issues, the support log will contain an entry similar to this:

2015-06-05T08:55:49.042+00:00 mgmt 2015-06-05 08:55:49,042 Level="INFO" Name="support.ldap" Message="Failed connecting to LDAP server" Address="server.example.com" Reasons="
ldaps://server.example.com : Can't contact LDAP server
ldap://server.example.com : Connect error"

The reason "**Connect error**" means that Pexip Infinity cannot verify the LDAP server's certificate.

Ensure that the LDAP server's TLS certificate (or the CA certificate that signed it, if it is not self-signed) is in the Pexip Infinity trust store (**Platform Configuration > Trusted CA Certificates**).

Connection errors: binding to the server fails e.g. invalid credentials

If Pexip Infinity can reach the configured LDAP server, but cannot connect to it due to binding errors, such as invalid credentials, the support log will contain an entry similar to this:

2015-06-05T09:11:03.765+00:00 mgmt 2015-06-05 09:11:03,765 Level="INFO" Name="support.ldap" Message="Failed connecting to LDAP server" Address="server.example.com" Reasons="
ldaps://server.example.com : Invalid credentials
ldap://server.example.com : Invalid credentials"

Ensure that you have entered the correct credentials. They should be for an enabled, non-expired, domain user service account (not the Administrator account), which has a password set to never expire. All usernames and passwords are case sensitive.

If you are certain that the account you are trying to bind with is configured correctly, try to bind using the:

- bare username of the service account (e.g. **ldapuser**)
- full DN of the service account (e.g. **CN=ldapuser,CN=Users,DC=example,DC=com**)
- Windows logon of the service account (e.g. **EXAMPLE\ldapuser**).

Connection errors when using insecure transport

You cannot specify the **LDAP server address** as an IP address if you have also selected the **Allow insecure transport** option. If the server address is not specified as an FQDN you will receive "Invalid credentials" error messages.

You cannot use an IP address because the authentication handshake is encrypted using SASL technology. To achieve this, various shared keys are used — things both sides know and use as part of the handshake but are not exchanged on the wire. In this case, it is the FQDN of the LDAP server that is used.

Therefore, if you need to use insecure transport, you must ensure that you refer to the LDAP server by its FQDN (and this is the hostname the server uses to identify itself, not just something that points to the IP address), so that the authentication will work. See Using ldapsearch or AD Explorer to view the LDAP database below for an example of how to discover an AD server's hostname.

Alternatively, you could use secure transport, referring to the LDAP server by any name that appears in its TLS certificate, and by loading all necessary trusted CA certificates onto Pexip Infinity.

Connection errors: Error syncing with LDAP

You can receive an "Error syncing with LDAP" error message when attempting to perform a VMR template synchronization.

This can be caused by invalid syntax in the template's **LDAP user filter** or **LDAP user search DN** fields. Check that all parentheses are balanced and are in the correct places, and that all operators are correctly positioned.

This message can also be received if you have not selected an LDAP sync source when configuring your VMR sync template.

Cannot log in to Pexip Infinity despite using correct credentials

If users receive a "Please enter the correct username and password for a staff account" message when trying to log in to Pexip Infinity, but they are using the correct username and password, this typically means that either:

- The LDAP server cannot be contacted:
  - These errors will be recorded in the Support log; see the connectivity troubleshooting guidelines above for more information.
- The LDAP server can be contacted but the correct user records are not being searched:
  - Check the Pexip Infinity LDAP configuration settings (**Users > User Authentication**) to ensure that all objectClass and LDAP field names have been spelled correctly, and that the base DN and user search DN fields contain the correct domain and organizational unit settings.
  - If you are using nested AD security groups, see Supporting nested security groups in Windows Active Directory.
- The LDAP server can be contacted and the user records can be found and authenticated, but the user is not authorized to access Pexip Infinity:
  - Check that LDAP roles have been configured on Pexip Infinity (**Users > LDAP Roles**).
  - Ensure that the user's LDAP account is associated with the LDAP group DNs / role combinations that are configured on Pexip Infinity.

Recovering local access

If necessary you can reinstate access via the Pexip Infinity local on-box database, so that users can log in via the default account (typically **admin**) and will have full administrator privileges. You may need to do this if, for example the **Authentication source** is configured as ***LDAP database*** and your connectivity to the LDAP server goes down or your credentials become invalid.

To reactivate your local admin account:

1. Log in to the Management Node over SSH.
2. For local admin access only, run the command:
   `authset LDAP LOCAL`
   or, for LDAP and local admin access, run the command:
   `authset LDAP BOTH`

You can also disable client certificate authentication so that you can log in to the Pexip Infinity Administrator interface via the standard login page.

To disable certificate-based authentication:

1. Log in to the Management Node over SSH.
2. Run the command:
   `authset CBA OFF`

If you forget the password for the Pexip Infinity Administrator interface, you can re-run the installation wizard, being sure to change only the **Web administration password** setting.

VMRs or devices not created as expected by a sync template

User search or user filters not being applied

If more or fewer VMRs or devices than expected (or no VMRs/devices at all) were created after performing a template synchronization, it is likely that the **LDAP base DN**, **LDAP user search DN** and **LDAP user filter** fields have been misconfigured.

Check that all objectCategory, objectClass and LDAP field names have been spelled correctly. Note that all LDAP user search and user filter contents are not case sensitive.

More information on Active Directory LDAP filtering can be found at http://social.technet.microsoft.com/wiki/contents/articles/5392.active-directory-ldap-syntax-filters.aspx.

Using ldapsearch or AD Explorer to view the LDAP database

Mac and Linux systems

You can use a command line tool such as **ldapsearch**, which is available for Mac and Linux systems, to help test and diagnose connectivity issues with the LDAP server. Note that **ldapsearch** is not installed on any Pexip Infinity nodes.

Here are some example **ldapsearch** queries you could use (after adapting the parameters as appropriate for your environment).

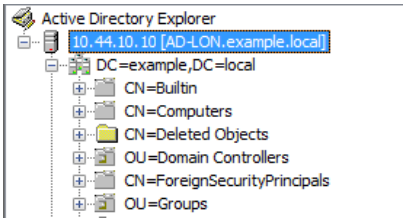
```
$ ldapsearch -v -h 10.0.0.8 -D "example\\admin123" -w password123 -b OU=people,DC=example,DC=com
```

This fetches the contents of OU (org unit) **people** from the LDAP server at **10.0.0.8** over TCP, binding as user (sAMAccountName) **admin123** in NetBIOS domain **example** with password **password123** using simple (insecure) authentication.

```
$ ldapsearch -v -h dc01.example.com -Y DIGEST-MD5 -U admin123 -w password123 -b
OU=people,DC=example,DC=com
```

This extends the previous example by addressing the LDAP server by its FQDN **dc01.example.com** and uses SASL/DIGEST-MD5 authentication.

Windows

Windows users can use Active Directory Explorer (AdExplorer) to navigate around and view AD structures and entries. See https://technet.microsoft.com/en-us/sysinternals/adexplorer for more information and links to download the software.

The example below shows how you can discover your AD server's actual hostname (**AD-LON.example.local** in this case) if you use AdExplorer to connect to your server via its IP address (**10.44.10.10** in this case):