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WebRTC Private Edition Guide

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Find links to all the topics in this guide.

Related documentation:

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- [For private edition](#)

WebRTC is a service available with the Genesys Multicloud CX private edition offering.

Overview

Learn more about WebRTC, its architecture, and how to support high availability and disaster recovery.

- About WebRTC
- Architecture
- High availability and disaster recovery

Configure and deploy

Find out how to configure and deploy WebRTC.

- Before you begin
- Configure WebRTC
- Configure Webphone
- Provision WebRTC
- Deploy
- Deploy Webphone
- Upgrade, rollback, or uninstall WebRTC

Observability

Learn how to monitor WebRTC with metrics, alerts, and logging.

- Observability
- Metrics and alerts
- Logging

About WebRTC

Contents

- [1 Supported Kubernetes platforms](#)

Learn about WebRTC and how it works in Genesys Multicloud CX private edition.

Related documentation:

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Web Real-Time Communication (WebRTC) is a real-time communication over the internet that enables an agent to connect with the Genesys contact center environment to perform their business operations.

WebRTC is a shared (multitenant) service that acts as the signaling and media gateway. The signaling gateway is used to interwork WebRTC with Session Initiation Protocol (SIP), and the media gateway is used to terminate the Interactive Connectivity Establishment (ICE) and Secure Real-time Transport Protocol (SRTP).

WebRTC bridges the calls that are initiated/received by the browser. The SIP Server handles these calls as a SIP call to provide core Genesys features such as routing and IVR. These features are handled by Genesys for browser endpoints with the help of MCP in the call flow. Third-party component CoTURN is used to implement TURN and STUN servers.

Supported Kubernetes platforms

WebRTC is supported on the following cloud platforms:

- Azure Kubernetes Service (AKS)
- Google Kubernetes Engine (GKE)

See the WebRTC Release Notes for information about when support was introduced.

Architecture

Learn about WebRTC architecture.

Related documentation:

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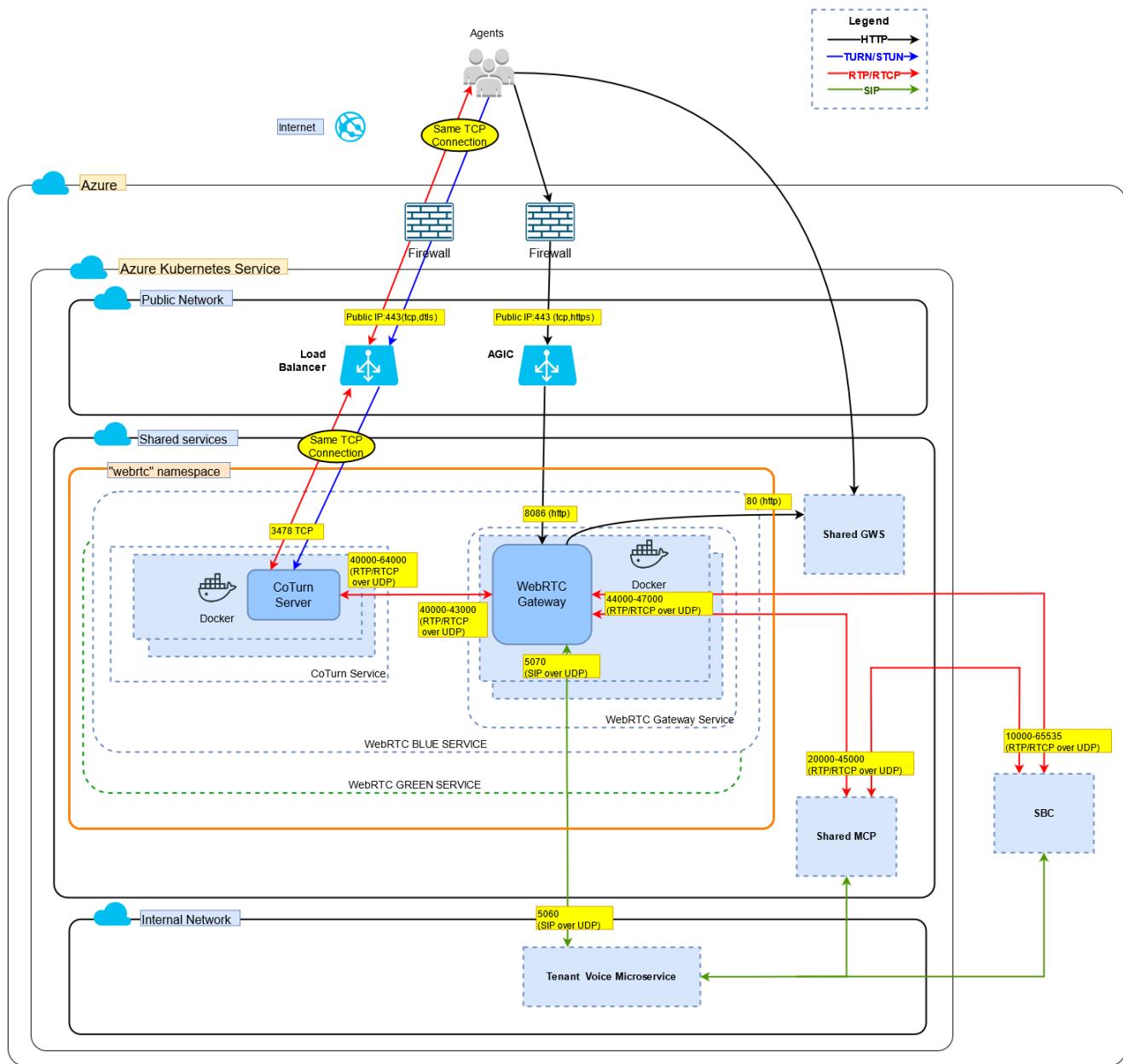
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For more information about the overall architecture of Genesys Private Edition Cloud, see:
Architecture.

Genesys Web Services (GWS) provides Tenant specific information to WebRTC. Workspace Web Edition (WWE) Agent Workspace retrieves all the required information such as tenant ID and WebRTC locations from GWS and sends them to WebRTC.

Architecture



High availability and disaster recovery

Find out how this service provides disaster recovery in the event the service goes down.

Related documentation:

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| Service | High Availability | Disaster Recovery | Where can you host this service? |
|----------------------|-------------------|-------------------|----------------------------------|
| WebRTC Media Service | $N = N (N+1)$ | Active-spare | Primary or secondary unit |

See High Availability information for all services: [High availability and disaster recovery](#)

Resources that are used for Primary region should be replicated for the backup region. For example, if a primary region has 10 WebRTC pods, the same number of pods must be deployed in the backup region also.

If a call from Genesys Voice Platform (GVP) /Session Border Controller (SBC) is from one region and WebRTC is in another region, the bandwidth between the regions is used for the backup. For this scenario, Pulse Code Modulation mu-law (PCMU) codec is used which requires 100kbps per call.

WebRTC will follow the Workspace Web Edition (WWE) pattern for its failover. For example, If WWE primary is location 1 and the backup is location 2, WebRTC will also backup from location 1 to location 2.

Before you begin

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Find out what to do before deploying WebRTC.

Related documentation:

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Limitations and assumptions

All prerequisites described under Third-party prerequisites, Genesys dependencies, and Secrets have been met.

Download the Helm charts

Download the Helm charts from the `webrtc` folder in the JFrog repository. See Helm charts and containers for WebRTC for the Helm chart version you must download for your release.

For information about how to download the Helm charts in Jfrog Edge, see the suite-level documentation: [Downloading your Genesys Multicloud CX containers](#)

WebRTC contains the following containers:

| Artifact | Type | Functionality | JFrog Containers and Helm charts |
|----------------|--------------------------|--|---|
| webrtc | webrtc gateway container | Handles agents' sessions, signalling, and media traffic. It also performs media transcoding. | https://webrtc/webrtc/ |
| coturn | coturn container | Utilizes TURN functionality | https://webrtc/coturn/ |
| webrtc-service | Helm chart | | https://webrtc-service-.tgz |

Third-party prerequisites

For information about setting up your Genesys Multicloud CX private edition platform, see [Software requirements](#).

The following are the third-party prerequisites for WebRTC:

| Third-party services | | | | |
|--|---------|---|--|--|
| Name | Version | Purpose | Notes | |
| Keda | 2.0 | Custom metrics for scaling. Use of Keda or HPA is configurable through Helm charts. | KEDA can be enabled or disabled for WebRTC. But, WebRTC cannot be configured to use HPA instead of KEDA. | |
| Load balancer | | VPC ingress. For NGINX Ingress Controller, a single regional Google external network LB with a static IP and wildcard DNS entry will pass HTTPS traffic to NGINX Ingress Controller which will terminate SSL traffic and will be setup as part of the platform setup. | | |
| A container image registry and Helm chart repository | | Used for downloading Genesys containers and Helm charts into the customer's repository to support a CI/CD pipeline. You can use any Docker OCI compliant registry. | | |
| Command Line Interface | | The command line interface tools to log in and work with the Kubernetes clusters. | | |

Storage requirements

WebRTC does not require persistent storage for any purposes except Gateway and CoTurn logs. The following table describes the storage requirements:

| Persistent Volume | Size | Type | IOPS | Functionality | Container | Critical | Backup needed |
|---------------------------|------|------|--------|---------------------------|-----------|----------|---------------|
| webrtc-gateway-log-volume | 50Gi | RW | medium | storing gateway log files | webrtc | Y | Y |

| | | | | | | | |
|--------------------------|------|----|--------|--------------------------|--------|---|---|
| webrtc-coturn-log-volume | 50Gi | RW | medium | storing coturn log files | coturn | N | Y |
|--------------------------|------|----|--------|--------------------------|--------|---|---|

Persistent Volume and Persistent Volume Claim will be created if they are configured. The size for them optional and should be adjusted according to log rate described below:

Gateway:

idle: 0.5 MB/hour per agent

active call: around 0.2MB per call per agent.

Example: For 24 full hours of work, where each agent call rate is constant and is around 7 to 10 calls per hour, we will require around ~500GB for 1000 agents, with around ~20GB being consumed per hour.

CoTurn:

For 1000 connected agents, the load rate is approximately 3.6 GB/hour which scales linearly and increases or decreases with the number of agents and stays constant whether calls are performed or not.

Network requirements

Ingress

WebRTC requires the following Ingress requirements:

- Persistent session stickiness based on cookie is mandatory. Stickiness cookie should contain the following attributes:
 - SameSite=None
 - Secure
 - Path=/
- No specific headers requirements
- Whitelisting (optional)
- TLS is mandatory

Secrets

WebRTC supports three types of secrets: CSI driver, Kubernetes secrets, and environment variables.

Important

GWS Secret for WebRTC should contain the following grants:

```
grant_type=authorization_code
grant_type=urn:ietf:params:oauth:grant-type:token-exchange
grant_type=urn:ietf:params:oauth:grant-type:jwt-bearer
grant_type=client_credentials
```

For GWS secrets, CSI or Kubernetes secret should contain gwsClient and gwsSecret key-values.

GWS secret for WebRTC must be created in the WebRTC namespace using the following specification as an example:

```
apiVersion: v1
kind: Secret
type: Opaque
metadata:
  name: webrtc-gws-secret
  namespace: webrtc
data:
  client_id: XXXXX
  client_secret: YYYYY
```

ConfigMaps

Not Applicable

WAF Rules

The following Web Application Firewall (WAF) rules should be disabled for WebRTC:

| WAF Rule | Number of rules |
|-------------------------------------|-----------------|
| REQUEST-920-PROTOCOL-ENFORCEMENT | 920300 |
| | 920440 |
| REQUEST-913-SCANNER-DETECTION | 913100 |
| | 913101 |
| REQUEST-921-PROTOCOL-ATTACK | 921150 |
| REQUEST-942-APPLICATION-ATTACK-SQLI | 942430 |

Pod Security Policy

Not applicable

Auto-scaling

WebRTC and CoTurn auto-scaling is performed by KEDA operator. The auto-scaling feature requires Prometheus metrics. To know more about KEDA, visit <https://keda.sh/docs/2.0/concepts/>.

Use the following option in YAML values file to enable the deployment of auto-scaling objects:

```
deployment:
  keda: true
```

You can configure the Polling interval and maximum number of replicas separately for Gateway pods and CoTurn pods using the following options:

```
gateway:
  scaling:
    pollingInterval: 30
    maxReplicaCount: 100

coturn:
  scaling:
    pollingInterval: 30
    maxReplicaCount: 100
```

- Gateway Pod Scaling
 - Sign-ins

```
gateway:
  scaling:
    pollingInterval: 30
    maxReplicaCount: 100
    prometheusAddress: http://monitoring-prometheus-prometheus.monitoring:9090
    thresholdSignins: 25
```

- CPU based scaling

WebRTC auto-scaling is also performed based on the CPU and memory usage. The following YAML shows how CPU and memory limits should be configured for Gateway pods in YAML values file:

```
gateway:
  scaling:
    prometheusAddress: http://monitoring-prometheus-prometheus.monitoring:9090
    pollingInterval: 30
    maxReplicaCount: 100
    thresholdSignins: 25
    thresholdCpu: 60
    thresholdMemory: 60
```

- CoTurn Pod scaling

Before you begin

Auto-scaling of CoTurn is performed based on CPU and memory usage only. The following YAML shows how CPU and memory limits should be configured for CoTurn pods in YAML values file:

```
coturn:
  scaling:
    pollingInterval: 30
    maxReplicaCount: 100
    thresholdCpu: 60
    thresholdMemory: 60
```

SMTP settings

Not applicable

Browser requirements

| Browsers | | |
|-------------------------|---|---|
| Name | Version | Notes |
| Firefox | Current release or one version previous | Genesys also supports the current ESR release. Genesys supports the transitional ESR release only during the time period in which the new ESR release is tested and certified. For more information, see Firefox ESR release cycle . Firefox updates itself automatically. Versions of Firefox are only an issue if your IT department restricts automatic updates. |
| Chrome | Current release or one version previous | Chrome updates itself automatically. Versions of Chrome are only an issue if your IT department restricts automatic updates. |
| Microsoft Edge Chromium | Current release | |

Genesys dependencies

WebRTC has dependencies on several other Genesys services and it is recommended that the provisioning and configuration of WebRTC be done after these services have been set up.

| Service | Functionality |
|---------|---------------|
|---------|---------------|

Before you begin

| | |
|---------------------|--|
| GWS | Used for environment and tenants configuration reading |
| GAuth | Used for WebRTC service and Agents authentication |
| GVP | Used for voice calls - conferences, recording, and so on |
| Voice microservice | Used to handle voice calls |
| Tenant microservice | Used to store tenant configuration |

For detailed information about the correct order of services deployment, see [Order of services deployment](#).

GDPR support

Not applicable

Configure WebRTC

Contents

- 1 Override Helm chart values
- 2 Configure Kubernetes
- 3 Configure security
 - 3.1 Arbitrary UIDs in AKS
- 4 Configure the service

Learn how to configure WebRTC.

Related documentation:

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RSS:

- [For private edition](#)

Override Helm chart values

Download the WebRTC Helm charts from JFrog using your credentials. Override the configuration parameters in the **values.yaml** file to provide deployment-specific values for certain parameters. You can override values in the Helm charts to configure Private Edition. For more information about overriding Helm chart values, see the "suite-level" documentation about how to override Helm chart values: [Overriding Helm chart values](#)

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|------------------------------|---|--------------|---------------|---------------|--|--|
| deployment.namespace | Name of Kubernetes namespace for WebRTC deployment | mandatory | webrtc | string | You can modify the default namespace used to deploy applications in the deployment.namespace option. | deployment: namespace: production |
| deployment.priorityClassName | Name of the priority class for pods that specify the importance of a pod relative to other pods | optional | | string | | |
| deployment.nodeSelector | Node selector for GStreamer and CoTurn pods | optional | | Specification | | deployment: nodeSelector: genesysengage.com/ |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|-------------------------------|--|--------------|---------------|---------------|--|--|
| | | | | | | nodepool: general |
| deployment.tolerations | Include this parameter in the Gateway and CoTurn, if the content of toleration exists. | optional | | Specification | | deployment: tolerations: - operator: Exists effect: NoSchedule key: "k8s.genesysengage.com/nodepool" |
| deployment.ingress.domain | Ingress domain | mandatory | | string | | deployment: ingress: domain: apps.vce-c0.eps.genesys.com |
| deployment.ingress.annotation | WebRTC Annotation for Ingress controller | mandatory | | Specification | As the default value of the HAProxy route timeout is set to 30 s, there is a possibility it interferes with the WebRTC long-polling timeout (30 s) and disconnect the session. | deployment ingress: annotations: kubernetes.io/ingress.class: nginx01-internal nginx.ingress.kubernetes.affinity: cookie nginx.ingress.kubernetes.affinity-mode: persistent nginx.ingress.kubernetes.ssl-redirect: "false" nginx.ingress.kubernetes.session-cookie-path: "/; Secure" nginx.ingress.kubernetes.session- |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|---------------------|---------------------------|--------------|---------------|---------------|-------|--|
| | | | | | | <pre>cookie-samesite: None {{!}}-{{!}}{{!}}deployment.ingress {{!}}{{!}}If this option is defined, tls option is declared in the Ingress specification {{!}}{{!}}optional {{!}}{{!}} {{!}}{{!}}Specification {{!}}{{!}} {{!}}{{!}}deployment: ingress: tls:</pre> <pre>secretName: webrtc.api01-eastus2.dev.tls-secret</pre> |
| deployment.affinity | Pod affinity descriptions | optional | | Specification | | <pre>deployment: affinity: podAntiAffinity: preferredDuringScheduling: - weight: 100 podAffinityTerm: labelSelector: matchExpressions: - key: servicename operator: In values: - webrtc-gateway - webrtc-coturn</pre> |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|-----------------------------|--|--------------|---------------|-------------------|--|---|
| | | | | | | topologyKey: failure-domain.beta.kubernetes.io/zone |
| deployment.dnsPolicy | Kubernetes DNS Policy applied in the Pods | optional | | | | deployment: nodeSelector: genesysengage.com/nodepool: general |
| deployment.dnsConfig | All DNS settings must be provided in the dnsConfig field in the Pod specification | optional | | | | deployment: dnsConfig: options: - name: ndots value: "3" |
| deployment.keda | Enable KEDA usage for the Gateway and CoTurn horizontal auto-scaling | optional | false | true/false | | |
| deployment.coturnDeployment | Type of CoTurn deployment - internal: the internal LBs are created and the IP addresses of that LBs must be exposed externally. - external: the external LBs are created with given external | mandatory | | internal/external | For Premise Edition - This parameter is configured as external | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|--------------------------------------|--|--------------|---------------|-------------|-------|--|
| | static IPs (IPs for the green and blue LBs must be set with <code>lbIpBlue</code> and <code>lbIpGreen</code> during the infra-color deployment.) | | | | | |
| deployment.coturnService.annotations | Annotation that is added to the <code>KubeServices annotation</code> LoadBalancer Service object | | | | | deployment: coturnService: annotations: service.beta.kubernetes.io/azure-load-balancer-resource-group: service-webrtc-westus2-dev |
| monitoring.enabled | Enable monitoring content - dashboards, alerts, metrics | optional | false | true/false | | |
| monitoring.dashboard | Enable ConfigMaps deployment that contains dashboards | optional | false | true/false | | |
| monitoring.prometheusMetrics | Enables Prometheus metrics to deploy PodMonitors | optional | false | true/false | | |
| monitoring.prometheusAlerts | Enable Prometheus rules for alerts | optional | false | true/false | | |
| image.imagePullSecrets | Secrets to use for image pull, list | mandatory | | | | image: imagePullSecrets: |

Configure WebRTC

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|--------------------------|---|--------------|---------------|---------------------|-------|---|
| | | | | | | - myRegistrySecret |
| image.pullPolicy | Kubernetes pull policy of all containers | optional | Always | Always/IfNotPresent | | |
| image.initContainerImage | Image for initialization container - used to create log folders. If image is not specified, the init container is not applied and the logs are written into logPath | optional | | string | | |
| image.webrtc | Repository/directory to get the Gateway image | mandatory | | string | | pureengage-docker-staging.jfrog.io/webrtc |
| image.coturn | Repository/directory to get the CoTurn image | mandatory | | string | | pureengage-docker-staging.jfrog.io/webrtc |
| image.webrtcVersion | Versions of the WebRTC Gateway container | mandatory | | string | | 9.0.000.88 |
| image.coturnVersion | Versions of the CoTurn container | mandatory | | string | | 9.0.000.88 |
| gateway.replicas | Number of Gateway pods on the deployment stage | optional | 1 | integer | | |
| gateway.workersCount | Number of Gateway worker threads that handle calls. 1 worker | optional | 3 | integer | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|-------------------------------|--|--------------|---------------|-----------------|-------|----------------------------------|
| | handles 25 registrations/calls. CPU and Memory request depends on the number of workers. | | | | | |
| gateway.voiceSipProxy | Voice microservice - SIP proxy address | mandatory | | string, address | | voice-sipproxy.voice.svc.cluster |
| gateway.turnExtCentrifugeLB | FQDNs of centrifuge LB | mandatory | | string, address | | |
| gateway.turnExtCentrifugGreen | FQDNs of centrifug green LB | mandatory | | string, address | | |
| gateway.authRedirectURI | GWS/WEE redirect URI for WEE authentication | mandatory | | string, address | | |
| gateway.authService | GAuth service address | mandatory | | string, address | | |
| gateway.envService | GWS9.x Environment service address | mandatory | | string, address | | |
| gateway.cfgService | GWS9.x configuration service address | optional | | string, address | | |
| gateway.enableTranscoding | Enable or disable transcoding on the Gateway side. Transcoding is enabled by default. If the transcoding is disabled, the Gateway can handle more agent sessions but OPUS codec is not | optional | true | true/false | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|--------------------------------|---|--------------|---------------|------------------|-------|---|
| | supported. | | | | | |
| gateway.enable1pccCalls | Specifies if the 1pcc operations are enabled | optional | false | true/false | | |
| gateway.arguments | Any additional options that are applied to the Gateway containers | optional | | Array of strings | | gateway: arguments: ['-codecs pcmu,pcma,opus=120', '-sip- disallowed- codecs opus,telephone- event'] |
| gateway.podAnnotations | Any additional annotations that are applied to the Gateway pods | optional | | | | gateway: podAnnotations: prometheus.io/ scrape: "true" prometheus.io/ port: "10052" prometheus.io/ path: "/metrics" |
| gateway.resources | Describes the resources requested for the Gateway pods. Important Do not specify this option, if you do not need resources requests/ limits. | optional | | Section | | gateway: resources: requests: cpu: 800 memory: 150 limits: memory: "8Gi" |
| gateway.resources.requests.cpu | Requested amount of CPU milliunits. | optional | 800 | integer | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|-----------------------------------|---|--------------|--|---|--|---------|
| | <p>Important This value is per worker and is multiplied by the gateway.workers option in helm</p> | | | | | |
| gateway.resources.requests.memory | Requested amount of Memory (in MB). | | | | | |
| gateway.resources.requests.memory | <p>Important This value is per worker and is multiplied by the gateway.workers option in helm</p> | optional | 150 | integer | | |
| gateway.resources.limits.memory | Absolute value for gateway.memory.memory usage limit | optional | "8Gi" | Kubernetes value for the resource limit | | |
| gateway.scaling | Describes the auto-scaling parameters. If the deployment.keda option is set to false, you can skip this option. | optional | | Section | <p>gateway: scaling: pollingInterval: 30 maxReplicaCount: 100 prometheusAddress: http://monitoring- prometheus- prometheus.monitoring:909 thresholdSignins: 70</p> | |
| gateway.scaling.prometheusAddress | Describes the auto-scaling parameters. If the deployment.keda option is set to false, you can skip | optional | http://monitoring- prometheus- prometheus.monitoring:909 | String, address | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|---------------------------------|--|--------------|---------------|--|-------|------------------------------------|
| | this option. | | | | | |
| gateway.scaling.pollingInterval | KEDA polling interval (in seconds) - the interval to check for scaling triggers. See KEDA documentation for more information. | optional | 30 | integer | | |
| gateway.scaling.maxReplicaCount | Maximum number of replicas that are raised by KEDA/HPA. See KEDA documentation for more information. | optional | 100 | integer | | |
| gateway.scaling.thresholdSign | In persons - number of registered agents that causes the Gateway auto-scaling if exceeded | optional | 71 | integer | | |
| gateway.budget.minAvailable | Option to configure the PodDisruptionBudget option. Do not specify this option if you do not need the PodDisruptionBudget option for the Gateway deployment. | optional | | Kubernetes PodDisruptionBudget (PBD) value | | gateway: budget: minAvailable: 50% |
| gateway.secrets.type | Describes where the secrets are taken - in Kubernetes secrets, CSI driver, or from the Environment | mandatory | | csi k8s env | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|---------------------------------|---|--------------|---------------|---------------|-------|---|
| | variables | | | | | |
| gateway.secrets.csi.gws | If the secrets.type option is set to csi, the name of the CSI object contains the GWS secret | | | string | | |
| gateway.secrets.k8s.gws | If the secrets.type option is set to k8s, the name of the Kubernetes Secret object that contains the GWS secret | | | string | | |
| gateway.secrets.env.gwsClient | If the secrets.type option is set to env, the value is GWS clientid created for WebRTC | | | string | | |
| gateway.secrets.value.gwsSecret | If the secrets.type option is set to env, the GWS secret for the client given clientid | | | string | | |
| gateway.securityContext | Security context for the Gateway container | optional | | Specification | | gateway: securityContext: runAsUser: 500 runAsGroup: 500 |
| gateway.serviceAccountName | Name of the ServiceAccount that is used to run the Gateway pod | optional | | string | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|-----------------|--|--------------|-------------------|-------------|-------|--|
| gateway.logPath | <p>Path to the log-directory, used for both - PVC or HostPath types of logs. Also, check the esServer option. If /mnt/log/webrtc is specified, the /mnt/log/webrtc//webrtcgw logfiles are created and used in the mentioned path. If the image.initContainerImage option is not specified, the folder with the pod name will not be created and the /mnt/log/webrtc/webrtcgw logfiles will be created.</p> <div style="border: 1px solid orange; padding: 5px; margin-left: 10px;"> Important If this option is set to stdout, the entire WebRTC GW logs are produced to the stdout in JSON format. </div> | mandatory | "/mnt/log/webrtc" | string | | "/export/vol1/PAT/infra/webrtc" |
| gateway.logPvc | Option for Persistent Volume Claim used for the Gateway logs. If logPvc is not defined, the | optional | | Section | | gateway: logPvc: pvcName: webtc-gateway-log-pvc |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|---------------------------|--|--------------|---------------|-------------|-------|---|
| | HostPath is used for the logs mount. | | | | | <pre> volumeName: webRTC-gateway-log-volume storageClassName: genesys-webrtc capacity: 5Gi volumeSpec: accessModes: - ReadWriteMany persistentVolumeReclaimPolicy: Retain nfs: path: /export/vol1/PAT/infra/webrtc server: 192.168.30.51 </pre> |
| gateway.logPvcName | Name of the Persistent Volume Claim. If this option is present, the PVCName is created. Else, the hostpath is used for the Gateway logs. | optional | | string | | |
| gateway.logPvc.volumeName | PersistentVolume name for the PVC. Single Volume is used for both green and blue deployments of the gateway | optional | | string | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|----------------------------------|--|--------------|---------------|-----------------------------|-------|--|
| gateway.logPvc.persistentVolume | If the Persistent Volume specification is configured in the gateway.logPvc.volumeSpec option, the gateway.logPvc.persistentVolume optional object with name from the gateway.logPvc.volumeName option is created using this specification. | optional | | Specification | | gateway: logPvc: volumeSpec: accessModes: - ReadWriteMany persistentVolumeReclaimPolicy: Retain nfs: path: /export/ vol1/PAT/ infra/ webrtc server: 192.168.30.51 |
| gateway.logPvc.volumeAnnotations | Any additional annotations that are used for the PersistentVolume if the gateway.logPvc.volumeSpec is specified here. | optional | | Specification | | gateway: logPvc: volumeAnnotations: pv.kubernetes.io/ bound-by-controller: 'yes' |
| gateway.esServerAddress | Specifies the destination for the ElasticSearch logging - ElasticSearch server address or stdout. Gateway produces messages in the ElasticSearch format. | optional | stdout | network address or "stdout" | | |
| gateway.restartPolicy | Restart policy for gateway pods. | Optional | Always | depends on cluster | | |
| coturn.port | Coturn port that is used by the | optional | 443 | integer | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|------------------------------|---|--------------|---------------|---------------|-------|--|
| | CoTurn Load Balancer | | | | | |
| coturn.lbIpBlue | External IP for CoTurn blue Load Balancer service. The coturn.lbIpBlue IP must be same as the one used for the gateway.turnExternalUriBlue A-record | mandatory | | IP address | | |
| coturn.lbIpGreen | External IP for CoTurn green Load Balancer service. The coturn.lbIpGreen must be same as the one used for the gateway.turnExternalUriGreen A-record | mandatory | | IP address | | |
| coturn.replicas | Number of CoTurn pods | optional | 1 | integer | | |
| coturn.podAnnotations | Any additional annotations that are applied for CoTurn pods | optional | | Specification | | coTURN: podAnnotations: pods/ realtime: "true" pods/ owner: "1051" |
| coturn.resources | Describes resources requested for the CoTurn pods. Do not specify this option if you do not need resources requests/limits. | optional | | Section | | coTURN: resources: requests: cpu: "0.5" memory: "768Mi" limits: memory: "8Gi" |
| coturn.resourceRequested.cpu | optional | | 0.5 | Kubernetes | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|---------------------------------|---|--------------|---------------|-------------------------------------|-------|---|
| | amount of CPU. Coturn requires 0.08CPU per call. | | | CPU request format | | |
| coturn.resource.requests.memory | Requested amount of memory | optional | 150 | Kubernetes memory request format | | |
| coturn.resource.limits.memory | Absolute value for the memory usage limit | optional | "8Gi" | Kubernetes value for resource limit | | |
| coturn.scaling | Describes the autoscaling parameters. If the deployment.keda option is set to false, you can skip this section | optional | | Section | | <pre> coturn: scaling: pollingInterval: 30 maxReplicaCount: 100 thresholdCpu: 60 thresholdMemory: 60 </pre> |
| coturn.scaling.pollingInterval | Specifies the KEDA polling interval in seconds - the interval to check each trigger on. Refer to KEDA documentation for more information. | optional | 30 | integer | | |
| coturn.scaling.maxReplicaCount | Maximum number of replicas that are raised by KEDA/HPA. Refer to KEDA documentation for more information. | optional | 100 | integer | | |
| coturn.scaling.thresholdSign | optional | | 71 | integer | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|--------------------------------|--|--------------|---------------|----------------------|-------|--|
| coturn.scaling.thresholdCpu | percentage In percentage. The target value is the average of the CPU resource metric across all the CoTurn Cpu pods, represented as a percentage of the requested value of the resource for the pods. | optional | 60 | integer | | |
| coturn.scaling.thresholdMemory | In percentage. The target value is the average of the memory resource metric across all the CoTurn Memory pods, represented as a percentage of the requested value of the resource for the pods. | optional | 60 | integer | | |
| coturn.budget.minAvailable | Option to configure PodDisruptionBudget. Do not specify this option, if you do not need PodDisruptionBudget for the CoTurn deployment. | optional | | Kubernetes PDB value | | coturn: budget: minAvailable: 50% |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|---------------------------|---|--------------|-------------------|---------------|-------|--|
| coturn.securityContext | Security context for the CoTurn container. | optional | | Specification | | coturn: securityContext: runAsUser: 500 runAsGroup: 500 |
| coturn.serviceAccountName | Name of the ServiceAccount the CoTurn pod. | optional | | string | | |
| coturn.logPath | Path to the log-directory. This can be the directory path or "stdout". This path is used for both PVC or HostPath types of logs. Example: If /mnt/log/webrtc is specified, "/mnt/log/webrtc//turn.xxx.log" logfile is created and used in the mentioned path. If image.initContainerImage is not specified, the folder with pod name will not be created and mnt/log/webrtc/turn.xxx.log logfile will be created. | mandatory | "/mnt/log/webrtc" | string | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|--------------------------------|--|--------------|-------------------|------------------------------------|-------|--|
| coturn.logPvc | Section for Persistent Volume Claim used for CoTurn logs. If this option not defined, the HostPath is used for logs mount. | optional | "/mnt/log/webrtc" | Section | | coturn: logPvc: pvcName: webrtc- coturn-log- pvc storageClassName: default capacity: 10Gi volumeName: webrtc- coturn-log- volume volumeSpec: nfs: server: 192.168.1.5 path: /storage/ webrtc volumeMode: Filesystem persistentVolumeReclaimPo Retain |
| coturn.logPvc.pvcName | Name of PersistentVolumeClaim. If this option is present, PVC will be created. Else, the HostPath is used for CoTurn logs. | optional | | string | | |
| coturn.logPvc.storageClassName | StorageClass name for the CoTurn PVC | optional | | string | | |
| coturn.logPvc.capacity | Volume capacity | optional | | Kubernetes capacity storage values | | |
| coturn.logPvc.volumeName | Persistent volume name for the | optional | | string | | |

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|---|--|--------------|---------------|--------------------|-------|---|
| | PVC. Single Volume is used for both green and blue deployments of the CoTurn logs | | | | | |
| coturn.logPvc.volumeSpec | If the Persistent Volume specification is configured in coturn.logPvc.volumeSpec, the Persistent Volume object with name from the coturn.logPvc.volumeName will be created using this specification. | optional | | Specification | | <pre> gateway: logPvc: volumeSpec: accessModes: - ReadWriteMany persistentVolumeReclaimPolicy: Retain nfs: path: /export/vol1/PAT/infra/webrtc server: 192.168.30.51 </pre> |
| coturn.logPvc.persistentVolumeAnnotations | Any additional annotations that are used for the Persistent Volume, if the coturn.logPvc.volumeSpec option is specified | optional | | Specification | | <pre> gateway: logPvc: volumeAnnotations: pv.kubernetes.io/bound-by-controller: 'yes' </pre> |
| coturn.restartPolicy | Restart policy for coturn pods. | optional | Always | depends on cluster | | |
| labels.common | Describes the additional labels for common resources | optional | | | | |
| labels.gateway | Describes the additional | optional | | | | |

Configure WebRTC

| Option name | Description | Is mandatory | Default value | Valid value | Notes | Example |
|---------------|--|--------------|---------------|-------------|-------|---------|
| | labels for the Gateway resources - pods, deployments, and services | | | | | |
| labels.coturn | Describes the additional labels for the CoTurn resources - pods, deployments, and services | optional | | | | |
| labels.alerts | Describes the additional labels for the alert objects | optional | | | | |

Configure Kubernetes

Document the layouts for the following so customers can create them if their Helm chart doesn't include a way to do this:

- *ConfigMaps*
- *Secrets*

Configure security

The security context settings define the privilege and access control settings for pods and containers.

By default, the user and group IDs are set in the **values.yaml** file as 500:500:500, meaning the **genesys** user.

```
securityContext:  
  runAsNonRoot: true  
  runAsUser: 500  
  runAsGroup: 500  
  fsGroup: 500
```

Arbitrary UIDs in AKS

If you want to use arbitrary UIDs in your Azure Kubernetes Services deployment, override the **securityContext** settings in the **values.yaml** file, so that you do not define any specific IDs.

```
podSecurityContext:  
  runAsNonRoot: true  
  runAsUser: null  
  runAsGroup: 0  
  fsGroup: null  
  
securityContext:  
  runAsNonRoot: true  
  runAsUser: null  
  runAsGroup: 0
```

Configure the service

Before proceeding with the deployment process, perform the following pre-steps:

1. **Review values-template.yaml in helm charts:** It provides all the available options with comments and explanations.
2. **Configure all the options in your own values file:** Configure/overwrite values for options that you need. Use the values-template.yaml file from the package that displays the list of available options with their description.

Important

Do not configure **deployment.type** and **deployment.color** options in values.yaml-file(s). These values should be used only during deployment process as command-line parameters to specify the deployment process.

Sample values.yaml file:

```
deployment:  
  namespace: webrtc  
  ingress:  
    domain: apps.vce-c0.eps.genesys.com  
    annotations:  
      kubernetes.io/ingress.class: nginx01-internal  
      nginx.ingress.kubernetes.io/affinity: cookie  
      nginx.ingress.kubernetes.io/affinity-mode: persistent  
      nginx.ingress.kubernetes.io/ssl-redirect: "false"  
      nginx.ingress.kubernetes.io/session-cookie-path: "/; Secure"  
      nginx.ingress.kubernetes.io/session-cookie-samesite: None  
    dnsPolicy: ClusterFirst  
    dnsConfig:  
      options:
```

```
- name: ndots
  value: "3"
keda: false
coturnDeployment: external

monitoring:
  enabled: false
  dashboards: false
  prometheusMetrics: false
  prometheusAlerts: false

image:
  imagePullSecrets:
    - webrtcjfrogsecret
  initContainerImage: pureengage-docker-staging.jfrog.io/alpine:3.7-curl
  webrtc: pureengage-docker-staging.jfrog.io/webrtc
  coturn: pureengage-docker-staging.jfrog.io/webrtc
  webrtcVersion: 9.0.000.88
  coturnVersion: 9.0.000.88

gateway:
  logPath: "/export/voll/PAT/infra/webrtc"
  logPvc:
    pvcName: webrtc-gateway-log-pvc
    volumeName: webrtc-gateway-log-volume
    storageClassName: genesys-webrtc
    capacity: 5Gi
  volumespec:
    accessModes:
      - ReadWriteMany
    persistentVolumeReclaimPolicy: Retain
    nfs:
      path: /export/voll/PAT/infra/webrtc
      server: 192.168.30.51
    esServer: stdout
    replicas: 1
    workersCount: 1
    voiceSipProxy: voice-siproxy.voice.svc.cluster.local:5080;transport=tcp
    turnExternalUriBlue: 192.168.30.208
    turnExternalUriGreen: 192.168.30.209
    authRedirectUri: http://gauth.apps.vce-c0.eps.genesys.com:80
    authService: http://gauth-auth.gauth.svc.cluster.local:80
    envService: https://gws.apps.vce-c0.eps.genesys.com
    resources:
      requests:
        # NB! 800m per worker, MUST be integer, not string - will be multiplied by
        workersCount in helm
        cpu: 800
        # NB! 150Mi per worker, MUST be integer, not string - will be multiplied by
        workersCount in helm
        memory: 150
      limits:
        memory: "8Gi"
    secrets:
      type: env
      env:
        gwsClient: external_api_client
        gwsSecret: secret
    securityContext:
      runAsUser: 500
```

```
runAsGroup: 500

coturn:
  logPath:          "/export/vol1/PAT/infra/coturn/"
  logPvc:
    pvcName:        webrtc-coturn-log-pvc
    volumeName:      webrtc-coturn-log-volume
    storageClassName: genesys-webrtc
    capacity:       5Gi
  volumeSpec:
    accessModes:
      - ReadWriteMany
  persistentVolumeReclaimPolicy: Retain
  nfs:
    path: /export/vol1/PAT/infra/webrtc
    server: 192.168.30.51
  replicas: 1
  port: 443
  lbIpBlue: 192.168.30.208
  lbIpGreen: 192.168.30.209
  securityContext:
    runAsUser: 500
    runAsGroup: 500
```

3. **PersistentVolume (PV) and PersistentVolumeClaim (PVC):** If you plan to use PV for logs, create the PV and then specify it for PVC of Gateway and CoTurn.

PV can also be created during the common-infrastructure deployment. You should review the `values-template.yaml` file and then configure the PV specification for Gateway and CoTurn.

Single PV/PVC pair will be used for both Green and Blue deployments of Gateway, and another single PV/PVC pair will be used for both Green and Blue deployments of CoTurn.

Configure Webphone

Contents

- 1 [Override Helm chart values](#)
- 2 [Configure the service](#)

Learn how to configure Webphone.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

Override Helm chart values

Download the Webphone Helm charts from JFrog using your credentials. Override the configuration parameters in the **values.yaml** file to provide deployment-specific values for certain parameters. You can override values in the Helm charts to configure Private Edition. For more information about overriding Helm chart values, see the "suite-level" documentation about how to override Helm chart values: [Overriding Helm chart values](#).

The following table includes the Helm chart values required for configuring Webphone service.

| Option Name | Description | Is Mandatory? | Default Value | Valid Value |
|------------------------|--|---------------|---------------|-------------|
| image.repository | Webphone image repository/directory. | mandatory | | string |
| image.tag | Version of Webphone container. | mandatory | | string |
| image.pullPolicy | Kubernetes pull policy of all containers. | mandatory | | string |
| image.imagePullSecrets | Secrets to pull image. | optional | | list |
| replicaCount | Number of desired Webphone pods. | optional | | integer |
| webphone.cfgService | GWS9.x environment service address for configuring Webphone service. | mandatory | | string |
| webphone.authService | GAAuth service address. | mandatory | | string |
| webphone.webrtcService | WebRTC service address. | mandatory | | string |

| Option Name | Description | Is Mandatory? | Default Value | Valid Value |
|--|---|---------------|--|-------------|
| webphone.webphoneServiceAddress. | Webphone service address. | mandatory | | string |
| webphone.authRedirectURLforWWE authentication. | GWS/WEE redirect URL for WWE authentication. | mandatory | | |
| secrets.gwsClientId | GWS client ID injected as environment variable. | mandatory | "webrtc_secret_client" | |
| k8s.gws | Name of Kubernetes Secret object that contains secret for GWS client ID. | optional | | |
| k8s.gwsSecretPath | File path in Kubernetes Secret object that contains the GWS secret. | optional | | |
| secrets.env.gws | Secret for GWS client ID, injected as environment variable. | optional | | string |
| service.enabled | To enable deployment of Kubernetes Service for Webphone pods. | mandatory | true | true/false |
| service.type | Type of Webphone Service object. | optional | ClusterIP | string |
| service.port | Port of Webphone Service object. | optional | 80 | integer |
| ingress.enabled | To enable deployment of Kubernetes Ingress for Webphone Ingress. This parameter can be enabled only if Webphone service is enabled. | mandatory | true | true/false |
| ingress.annotations | Any additional annotations applied for Webphone Ingress. | optional | ingress: annotations: nginx.ingress.kubernetes.io/ affinity: cookie nginx.ingress.kubernetes.io/ affinity-mode: persistent | |

| Option Name | Description | Is Mandatory? | Default Value | Valid Value |
|--------------------------|--|---------------|--|-------------|
| | | | nginx.ingress.kubernetes.io/session-cookie-name: webphonesession nginx.ingress.kubernetes.io/session-cookie-path: /; Secure nginx.ingress.kubernetes.io/session-cookie-secure: "true" nginx.ingress.kubernetes.io/session-cookie-samesite: None | |
| ingress.host | FQDNs of Webphone. This field is mandatory, if ingress is enabled. | optional | | string |
| ingress.ingressClassName | Classname of ingress. | optional | | string |
| resources | Resources requested for Webphone pods. | optional | resources: limits: cpu: 500m memory: 1024Mi requests: cpu: 50m memory: 128M | list |
| podAnnotations | Any additional annotations applied for Webphone pods. | optional | | array |
| podSecurityContext | Security context for Webphone pod. | optional | | array |
| securityContext | Security context for Webphone container. | optional | | array |
| nodeSelector | Node selector for Webphone pods. | optional | | array |
| tolerations | If toleration exists, this parameter is inserted into toleration of Webphone pods. | optional | | array |
| affinity | Pod affinity descriptions. | | | |

| Option Name | Description | Is Mandatory? | Default Value | Valid Value |
|---|--|---------------|---------------|-------------|
| serviceAccountName | Name of the Service Account used to run the Webphone pods. | optional | default | |
| priorityClassName | Name of the priority class for pods that indicates the importance of a pod relative to other pods. | optional | | string |
| dnsPolicy | Kubernetes DNS Policy applied in the pods. | optional | | |
| dnsConfig | All DNS settings are provided using this field in the Pod Spec. | optional | | |
| autoscaling.enabled | To enable autoscaling of Webphone. | mandatory | false | true/false |
| autoscaling.minReplicas | Minimal number of Webphone pods. | optional | 1 | integer |
| autoscaling.maxReplicas | Maximum number of Webphone pods. | optional | 2 | integer |
| autoscaling.targetCPUUtilizationPercentage | CPU usage threshold to trigger autoscaling. | optional | 80 | integer |
| autoscaling.targetMemoryUtilizationPercentage | Memory usage threshold to trigger autoscaling. | optional | | integer |
| monitoring.enabled | To enable monitoring of Webphone service. | mandatory | false | true/false |
| monitoring.scrapeInterval | | optional | | |

Configure the service

Before proceeding with the deployment process, perform the following pre-steps:

- 1. Review values-template.yaml in helm charts:** It provides all the available options with comments and explanations.
- 2. Configure all the options in your own values file:** Configure/overwrite values for options that you need. Use the values **template.yaml** file from the package that displays the list of available options

with their description.

Sample values.yaml file:

```
image:
  repository: pureengageuse1-docker-multicloud.jfrog.io
  tag: 100.0.007.0000
  pullPolicy: IfNotPresent

webphone:
  cfgService: "https://gws."
  authService: "https://gauth."
  webrtcService: "https://webrtc."
  webphoneService: "https://webphone."

secrets:
  gwsClientId: "webrtc_service_client"
  env:
    gws: "secret"

ingress:
  annotations:
    kubernetes.io/ingress.class: nginx
    kubernetes.io/ingress.allow-http: "true"
    nginx.ingress.kubernetes.io/ssl-redirect: "false"
    nginx.ingress.kubernetes.io/affinity: cookie
    nginx.ingress.kubernetes.io/affinity-mode: persistent
    nginx.ingress.kubernetes.io/session-cookie-name: webphonesession
    nginx.ingress.kubernetes.io/session-cookie-path: /; Secure
    nginx.ingress.kubernetes.io/session-cookie-secure: "true"
    nginx.ingress.kubernetes.io/session-cookie-samesite: None
    nginx.ingress.kubernetes.io/session-cookie-conditional-samesite-none: "true"
    nginx.ingress.kubernetes.io/session-cookie-max-age: "86400"
    nginx.ingress.kubernetes.io/session-cookie-change-on-failure: "true"
  hosts:
    - host: webphone.apps.qrtp6qa.westus2.aroapp.io
      paths:
        - path: "/"

monitoring:
  enabled: false

serviceAccount:
  create: false
  name: default

podSecurityContext:
  runAsGroup: 0
  runAsNonRoot: true

rbac:
  create: false
```

Provision WebRTC

Contents

- [1 Tenant provisioning](#)

- Administrator

WebRTC does not require tenant provisioning.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

List any provisioning needed to deploy, run, or manage the service. For example:

- *Designer: Create an Access Group specific to Designer Developer, Admin.*
- *Agent Setup: Create Agent Setup options to provide access to Administrator, Supervisor, or Ops.*
- *Genesys Info Mart: Update the CTL_CONFIG table in the GIM DB to control ETL and DB maintenance behavior.*

Tenant provisioning

Describe how to provision the tenant service for .

Deploy

Contents

- 1 Assumptions
- 2 Deploy
 - 2.1 Deploying WebRTC using internal CoTurn Load Balancer
 - 2.2 Deployment with external CoTurn Load Balancer
 - 2.3 Cutover
- 3 Validate the deployment

Learn how to deploy WebRTC Media Service (WebRTC) into a private edition environment.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

Assumptions

- The instructions on this page assume you are deploying the service in a service-specific namespace, named in accordance with the requirements on [Creating namespaces](#). If you are using a single namespace for all private edition services, replace the namespace element in the commands on this page with the name of your single namespace or project.
- Similarly, the configuration and environment setup instructions assume you need to create namespace-specific (in other words, service-specific) secrets. If you are using a single namespace for all private edition services, you might not need to create separate secrets for each service, depending on your credentials management requirements. However, if you do create service-specific secrets in a single namespace, be sure to avoid naming conflicts.

Important

Make sure to review [Before you begin](#) for the full list of prerequisites required to deploy WebRTC.

WebRTC uses blue-green model of deployment. It has the following main deployment principles:

- Both components - WebRTC Gateway and CoTurn Server - are deployed for each color and switched together
- Blue WebRTC Gateway is always configured to work with Blue CoTurn and green WebRTC Gateway is always configured to work with green CoTurn
- WebRTC have two FQDNs to reach active and inactive deployments:
 - **webrtc.domain.com** - active deployment. For example: webrtc.genesyshtcc.com
 - **webrtc-test.domain.com** - inactive deployment for tests. For example: webrtc-test.genesyshtcc.com

Deploy

You can deploy WebRTC using:

- Internal CoTurn Load Balancer or
- External CoTurn Load Balancer

Deploying WebRTC using internal CoTurn Load Balancer

Initial deployment and Upgrade use the same sequence:

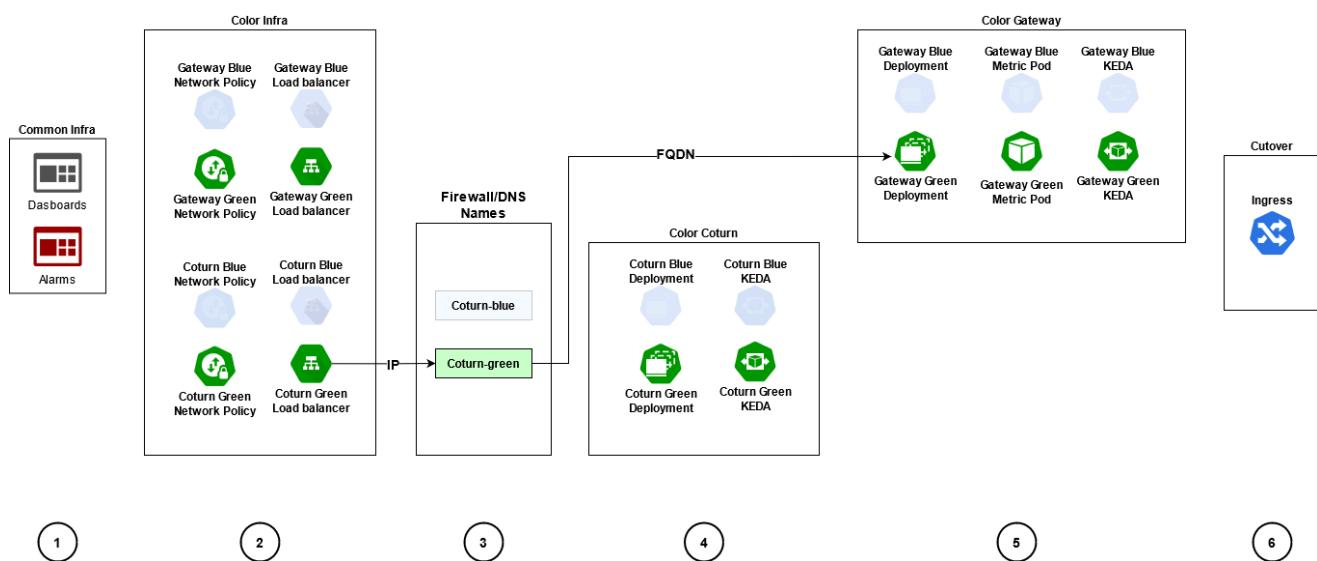
1. Deploy/upgrade inactive color of deployment
2. Make the cutover

You need to deploy the Color Infra package with CoTurn Load Balancer to get the IP address assigned automatically for the CoTurn Load Balancer by the infrastructure. Then, the infrastructure team should assign the IP to the CoTurn Load Balancer, create the FQDN for the IP and ensure that the IP is set in the firewall and is available from outside the cluster.

Important

The IP address assigned to the CoTurn Load Balancer must be external and available outside the cluster. Else, the media will not get through the WebRTC.

The following image shows the steps involved in deploying WebRTC using the internal CoTurn Load Balancer:



Follow the below steps to deploy WebRTC using internal CoTurn Load Balancer:

1. **Create common infrastructure elements such as dashboards and alarms:** This step deploys dashboards, alarms, and other common infrastructure elements.

Important

You should perform this step even if you do not require the dashboard and alarms.

Run the following command to create the common infrastructure elements:

```
helm upgrade --install -f {Webrtc Values files} --set-string deployment.type=infra --set-string deployment.color="" webrtc-infra {HelmRepoPath}/webrtc-service --version={WebRTC Charts Version}
```

Example:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=infra --set-string deployment.color="" webrtc-infra wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

2. **Create infrastructure elements for the deployment color:** This step deploys the infrastructure objects such as Turn Load Balancer, Gateway Service Object, Gateway Network Policies, and Turn Network Policies for the given color of deployment.

You should also specify the INACTIVE color of deployment in this step.

Important

You should configure the deployment.coturnDeployment option with the value internal in your values.yaml file.

Run the following command to deploy the infrastructure objects:

```
helm upgrade --install -f {Webrtc Values files} --set-string deployment.type=infra --set-string deployment.color={INACTIVE_COLOR} webrtc-infra-{INACTIVE_COLOR} {HelmRepoPath}/webrtc-service --version={WebRTC Charts Version}
```

Example:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=infra --set-string deployment.color=blue webrtc-infra-blue wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

3. **Get the IPs from the CoTurn Load Balancers, create DNS records and firewall rules:** This step gets the IP address from the CoTurn Load Balancer created in Step 2. The name of LoadBalancer will be similar to: webrtc-coturn-service-{COLOR}.

Create appropriate FQDN for this IP address in your DNS. This FQDN will be used by the WebRTC agents from outside the cluster to establish the RTP stream. Though you can use the IP address as it is, it is not the best practice to do so.

4. **Create CoTurn elements for the deployment color:** This step is to Upgrade/Deploy CoTurn for INACTIVE color.

Run the following command to upgrade/deploy the INACTIVE color of deployment:

```
helm upgrade --install -f {WebRTC Values files} --set-string deployment.type=coturn --set-string deployment.color={INACTIVE_COLOR} webrtc-coturn-{INACTIVE_COLOR} {HelmRepoPath }/webrtc-service --version={WebRTC Charts Version}
```

Example:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=coturn --set-string deployment.color=blue webrtc-coturn-blue wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

5. **Create Gateway elements for deployment color using the information from Step 3:** This step is to Upgrade/Deploy Gateway for INACTIVE color. You should also specify the external FQDN of the CoTurn LoadBalancer in this step using the gateway.turnExternalUriBlue or gateway.turnExternalUriGreen options.

Run the following command:

```
helm upgrade --install -f {WebRTC Values files} --set-string deployment.type=gateway --set-string deployment.color={INACTIVE_COLOR} --set-string gateway.turnExternalUri{INACTIVE_COLOR}={COTURN FQDN INACTIVE_COLOR} webrtc-gateway-{INACTIVE_COLOR} {HelmRepoPath }/webrtc-service --version={WebRTC Charts Version}
```

Example for Blue deployment:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=gateway --set-string deployment.color=blue --set-string gateway.turnExternalUriBlue=turn-blue.ext.mydomain.com webrtc-gateway-blue wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

Or, you can specify the IP of the Blue CoTurn Load Balancer

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=gateway --set-string deployment.color=blue --set-string gateway.turnExternalUriBlue=12.106.34.55 webrtc-gateway-blue wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

6. **Create/update Ingress controller rules for Active/Inactive routing for Gateway deployments:** This step is to Install/upgrade ingress without changing the active color. The same step is used for the Cutover.

Important

If you are deploying/upgrading green, specify the current ACTIVE color of deployment in the deployment.color option. Then specify blue and vice versa. If you are deploying/upgrading green and specify green for the **cutover** step, the current active deployment will be switched to the just deployed/upgraded green.

You must perform this step even if you are not planning to make the cutover right now. This step is to upgrade the ingress and environment.

Run the following command to create/upgrade Ingress controller rules:

```
helm upgrade --install -f {WebRTC Values files} --set-string deployment.type=cutover --set-string deployment.color={ACTIVE_COLOR} webrtc-ingress {HelmRepoPath }/webrtc-service --version={WebRTC Charts Version}
```

Example:

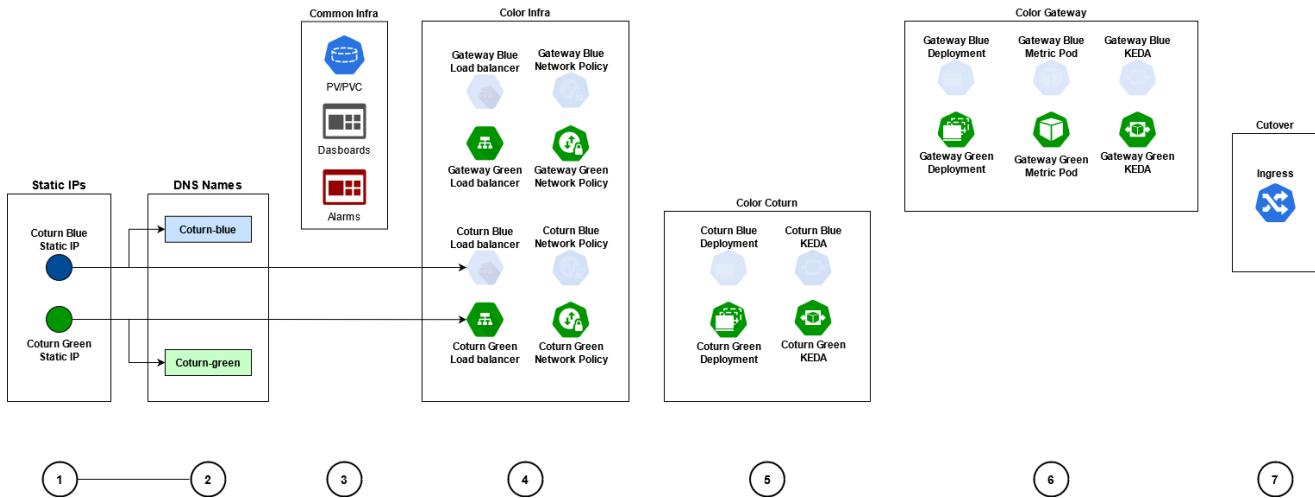
```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=cutover --set-string deployment.color=green webrtc-ingress wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

Deployment with external CoTurn Load Balancer

Initial deployment and Upgrade use the same sequence:

1. Deploy/upgrade inactive color of deployment
2. Make the cutover

The following image shows the steps involved in deploying WebRTC using the external CoTurn Load Balancer:



Follow the below steps to deploy WebRTC with external CoTurn Load Balancer

1. **Create static IPs for CoTurn:** This step is to specify the pre-created public IP for CoTurn Green in the `coturn.lbIpGreen` option and public IP for CoTurn Blue in the `coturn.lbIpBlue` option.
2. **Create DNS records for the created IPs:** This step is to specify the public FQDNs for CoTurn. Specify the pre-created public FQDN for CoTurn Green in the `gateway.turnExternalUriGreen` option and public FQDN for CoTurn Blue in the `gateway.turnExternalUriBlue` option.
3. **Create common infrastructure elements:** This step will deploy Persistent Volumes, Persistent Volume Claims, dashboards, alarms, and other common infrastructure elements.

Important

You need to run this step even if you are not using the dashboard and alarms.

Run the following command to create the infrastructure elements:

```
helm upgrade --install -f {WebRTC Values files} --set-string deployment.type=infra --set-string deployment.color="" webrtc-infra {HelmRepoPath}/webrtc-service --version={WebRTC Charts Version}
```

Example:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=infra --set-
```

```
string deployment.color="" webrtc-infra wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

4. **Create infrastructure elements for deployment color:** This step is to deploy the infrastructure objects such as Turn Load Balancer, Gateway Service Object, Gateway Network Policies, and Turn Network Policies for the given color of deployment.

You must specify INACTIVE color of deployment for this step.

Important

Configure the deployment.coturnDeployment option with the value external in your values.yaml file.

Run the following command to create the infrastructure elements:

```
helm upgrade --install -f {Webrtc Values files} --set-string deployment.type=infra --set-string deployment.color={INACTIVE_COLOR} webrtc-infra-{INACTIVE_COLOR} {HelmRepoPath }/webrtc-service --version={WebRTC Charts Version}
```

Example:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=infra --set-string deployment.color=blue webrtc-infra-blue wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

5. **Create CoTurn elements for deployment color:** This step is to upgrade/deploy CoTurn for inactive color.

Run the following command to specify the INACTIVE color of deployment:

```
helm upgrade --install -f {Webrtc Values files} --set-string deployment.type=coturn --set-string deployment.color={INACTIVE_COLOR} webrtc-coturn-{INACTIVE_COLOR} {HelmRepoPath }/webrtc-service --version={WebRTC Charts Version}
```

Example:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=coturn --set-string deployment.color=blue webrtc-coturn-blue wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

6. **Create Gateway elements for deployment color:** This step is to upgrade/deploy the Gateway for inactive color.

Important

CoTurn DNS name is used for Gateway deployment as a parameter in the corresponding values.yaml file.

Run the following command to specify the INACTIVE color of deployment:

```
helm upgrade --install -f {Webrtc Values files} --set-string deployment.type=gateway --set-string deployment.color={INACTIVE_COLOR} webrtc-gateway-{INACTIVE_COLOR} {HelmRepoPath }/webrtc-service --version={WebRTC Charts Version}
```

Example:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=gateway --set-string deployment.color=blue webrtc-gateway-blue wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

7. **Create/update Ingress controller rules for Active/Inactive routing for the Gateway deployments:** This step is to install/upgrade ingress without changing the active color. The same step is also used for the Cutover.

Important

If you are deploying/upgrading green, specify the current ACTIVE color of deployment in the deployment.color option which is blue and vice versa. If you deploying/upgrading green and specify green for the **cutover** step, the current active deployment will be switched to the just deployed/upgraded green.

Important

You must perform this step even if you do not plan to make cutover right now. This step is to upgrade the ingress and environment.

Run the following command to create/upgrade Ingress controller rules:

```
helm upgrade --install -f {WebRTC Values files} --set-string deployment.type=cutover --set-string deployment.color={ACTIVE_COLOR} webrtc-ingress {HelmRepoPath}/webrtc-service --version={WebRTC Charts Version}
```

Example:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=cutover --set-string deployment.color=green webrtc-ingress wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

Cutover

During cutover, it switches active color of deployment. This step should be performed only after you confirm that the newly installed/upgraded deployment is alive and functional. You must specify the current INACTIVE color of deployment in the deployment.color option - deployment that was just deployed/upgraded and tested. Run the following command to specify the cutover:

```
helm upgrade --install -f {WebRTC Values files} --set-string deployment.type=cutover --set-string deployment.color={INACTIVE_COLOR} webrtc-ingress {HelmRepoPath}/webrtc-service --version={WebRTC Charts Version}
```

Example:

```
helm upgrade --install -f ./k8s/values.yaml --set-string deployment.type=cutover --set-string deployment.color=blue webrtc-ingress wrtchelmrepodevwestus2/webrtc-service --version=0.1.93 -n webrtc
```

Important

You need to use PersistentVolume and PersistentVolumeClaim instead of HostPath logs of Gateway pods and CoTurn Pods.

Validate the deployment

Follow the given steps to validate the deployment.

1. Verify PVCs are created and bound

```
kubectl get pvc
```

Sample output:

| NAME | STORAGECLASS | STATUS | VOLUME | CAPACITY | ACCESS |
|------------------------|----------------|--------|---------------------------|----------|--------|
| webrtc-coturn-log-pvc | | Bound | webrtc-coturn-log-volume | 5Gi | |
| webrtc-gateway-log-pvc | genesys-webrtc | Bound | webrtc-gateway-log-volume | 5Gi | |
| | | | | | |

2. Validate CoTurn and Gateway services

```
kubectl get svc
```

Sample output:

| NAME | PORT(S) | TYPE | CLUSTER-IP | EXTERNAL-IP |
|-----------------------------|---------------|--------------|---------------|-----------------|
| webrtc-coturn-service-blue | 443:31457/TCP | LoadBalancer | 10.202.51.156 | 192.168.30.208 |
| webrtc-gateway-service-blue | 67m | ClusterIP | 10.202.47.170 | 80/TCP,8080/TCP |

3. Query pods in the WebRTC namespace to confirm that pod is created, and in running status

```
kubectl get pods
```

Sample output:

| NAME | READY | STATUS | RESTARTS | AGE |
|-------------------------------------|-------|---------|----------|-------|
| webrtc-coturn-blue-b5db74c96-mh9jv | 1/1 | Running | 0 | 4m20s |
| webrtc-gateway-blue-d7ff45677-vbdg9 | 1/1 | Running | 0 | 86s |

4. Validate Ingress configuration

```
kubectl get ingress
```

Sample output:

| NAME | HOSTS | CLASS |
|--------------------|-------|--|
| webrtc-ingress-int | | webrtc.apps.vce-c0.eps.genesys.com,webrtc-test.apps.vce- |

```
c0.eps.genesys.com      80      68s
```

5. Validate Ingress Edge route configuration

```
kubectl get route
```

Sample output:

| NAME | HOST/PORT | PORT | TERMINATION | WILDCARD | PATH |
|-----------------------------|---|------|-------------|----------|-------|
| webrtc-gateway-service-blue | webrtc.apps.qrtpf6qa.westus2.aroapp.io | | | | |
| webrtc-gateway-service-blue | web edge | | None | | / |
| webrtc-ingress-int-cvdtt | webrtc.apps.qrtpf6qa.westus2.aroapp.io | | | | / |
| webrtc-gateway-service-blue | web | | None | | /blue |
| webrtc-ingress-int-trcvh | webrtc.apps.qrtpf6qa.westus2.aroapp.io | | | | /blue |
| webrtc-gateway-service-blue | web | | None | | /blue |
| webrtc-ingress-int-wf6x9 | webrtc-test.apps.qrtpf6qa.westus2.aroapp.io | | | | /blue |
| webrtc-gateway-service-blue | web | | None | | |

6. Query Ingress for made available WebRTC Web API

```
kubectl get ingress
```

Copy the WebRTC API from the Ingress output:

Sample output:

| NAME | CLASS |
|--------------------|--|
| HOSTS | |
| ADDRESS | PORTS AGE |
| webrtc-ingress-int | webrtc.apps.vce-c0.eps.genesys.com,webrtc-test.apps.vce- |
| c0.eps.genesys.com | 80 3h26m |

Curl WebRTC "ping" API:

```
curl -s webrtc.apps.vce-c0.eps.genesys.com/ping
{"state":"up","version":"9.0.000.89","path":"blue"}
```

Deploy Webphone

Contents

- [1 Prerequisites](#)
- [2 Deploy Webphone](#)

Learn how to deploy Webphone.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

Prerequisites

You must deploy the following services before deploying Webphone:

1. GWS
2. GAuth
3. GVP
4. Voice Microservice
5. Tenant Microservice
6. WebRTC

GWS Secret for Webphone must contain the following grants:

```
grant_type=authorization_code
grant_type=urn:ietf:params:oauth:grant-type:token-exchange
grant_type=urn:ietf:params:oauth:grant-type:jwt-bearer
grant_type=client_credentials
```

Deploy Webphone

Run the following command to deploy Webphone with single run of Helm.

```
helm upgrade --install -f override_values webphone-service wrtchelmrepo/webphone --
version=100.0.007+0003 -n webrtc
```

Upgrade, rollback, or uninstall WebRTC

Contents

- [1 Upgrade WebRTC](#)
- [2 Rollback WebRTC](#)
- [3 Uninstall WebRTC](#)
 - [3.1 Uninstall Ingress](#)

Learn how to upgrade, rollback or uninstall WebRTC.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

Upgrade WebRTC

Follow the same process explained in the Deploy section for the upgrade.

Rollback WebRTC

WebRTC uses green-blue upgrade model. This means that the upgrade is performed for the currently inactive color of deployment, and then the cutover is performed. After the cutover, the new version will be active and the previous version gets inactive. To perform a rollback to the previous version, make a cutover. This makes the previous version active.

Uninstall WebRTC

This section describes the steps involved in uninstalling WebRTC.

Uninstall Ingress

1. Run the following helm command to remove the ingress configuration:

```
helm delete webrtc-ingress -n webrtc
```

Validate

Run the following command to validate if the ingress configuration is removed.

```
kubectl get ingress -n webrtc
```

2. Uninstall Gateway deployment

Run the following helm command to remove the Gateway deployment:

```
helm delete webrtc-gateway-{color} -n webrtc
```

Important

Update color for blue/green requirements.

Validate

Run the following command to validate if the Gateway deployment is removed:

```
kubectl get pods -n webrtc
```

3. Uninstall CoTurn deployment

Run the following helm command to remove the CoTurn deployment:

```
helm delete webrtc-coturn-{color} -n webrtc
```

Validate

Run the following command to validate if the CoTurn deployment is removed:

```
kubectl get pods -n webrtc
```

4. Uninstall CoTurn and Gateway services

Run the following helm command to uninstall CoTurn and Gateway services:

```
helm delete webrtc-infra-{color} -n webrtc
```

Validate

Run the following command to validate if the CoTurn and Gateway services are removed:

```
kubectl get pods -n webrtc
```

5. Uninstall Persistent Volumes (PV) and Persistent Volume Claims (PVC)

Run the following helm command to uninstall Persistent Volumes and Persistent Volume Claims:

```
helm delete webrtc-infra -n webrtc
```

Important

This is optional if you want to keep PVs and PVCs.

Validate

Run the following command to validate if the PVs and PVCs are removed:

```
kubectl get pv -l service=webrtcoc get pvc -n webrtc
```

Configure WebRTC Agents with Genesys Softphone

Contents

- [1 Pre-requisites](#)
- [2 Configure WebRTC agent with Genesys Softphone using Agent Setup](#)
- [3 Configure Genesys Softphone in WebRTC mode](#)
 - [3.1 Enabling Dynamic Configuration Connector in Connector Mode](#)

Configure WebRTC agents with Genesys Softphone in Genesys cloud using Agent Setup.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

Pre-requisites

- Deploy WebRTC in the Tenant's region. For more information, see [Configure and Deploy WebRTC](#).
- Install Genesys Softphone in the Agent's place. For more information, see [Deploying Genesys Softphone](#).
- Install WWE 9.0+ version, which is a part of GWS.
- Install GWS 9.0+ version in the tenant's region.

Genesys Softphone with WWE works in a Connector mode. In the Connector mode, the configuration is retrieved from the Configuration Server by WWE and sent to the Genesys Softphone.

To configure the Genesys Softphone in WebRTC mode, see [Genesys Softphone in WebRTC mode](#).

Configure WebRTC agent with Genesys Softphone using Agent Setup

To configure WebRTC agent with Genesys Softphone using Agent Setup:

1. Log in to the **Agent Setup** for the corresponding tenant.
2. Select **Users**.
3. Click **Add User**.
4. Update all the required fields and configure the phone number.
5. Select the **Softphone** check box and **Genesys SoftPhone with WebRTC** in the **SIP Phone Type** drop-down to assign all the mandatory DN level options.
6. Click **Save**.

7. Go to **Users > Annex** section and click **Add Section**.
8. In the **Group name** text box, type interaction-workspace and click **+** to configure the options. Use the following table to fill the options and its values.

| Option name | Option value for WVE9 |
|--|---|
| privilege.sipendpoint.can-use | true |
| privilege.webrtc.can-use | false |
| privilege.voice.can-use | true |
| sipendpoint.transport-protocol | https |
| sipendpoint.uri | https://localhost:8000 |
| sipendpoint.enable-webrtc-auth | true |
| sipendpoint.codecs.enabled.audio | opus |
| sipendpoint.enable-webrtc-signin-with-switchname | true |
| | {http_proxy_fqdn:http_proxy_port} |
| sipendpoint.proxies.proxy0.http_proxy | Example: sipendpoint.proxies.proxy0.http_proxy = proxy.company.com:8080 |
| webrtc.server-urn | <p>webrtc.service-urn = webrtc.service-urn can be templated using "expression.gws-url.capturing-groups" option.</p> <p>This option can be configured on CloudCluster application level and/or Agent Group level. Hence, the customer need not configure it for each agent.</p> |

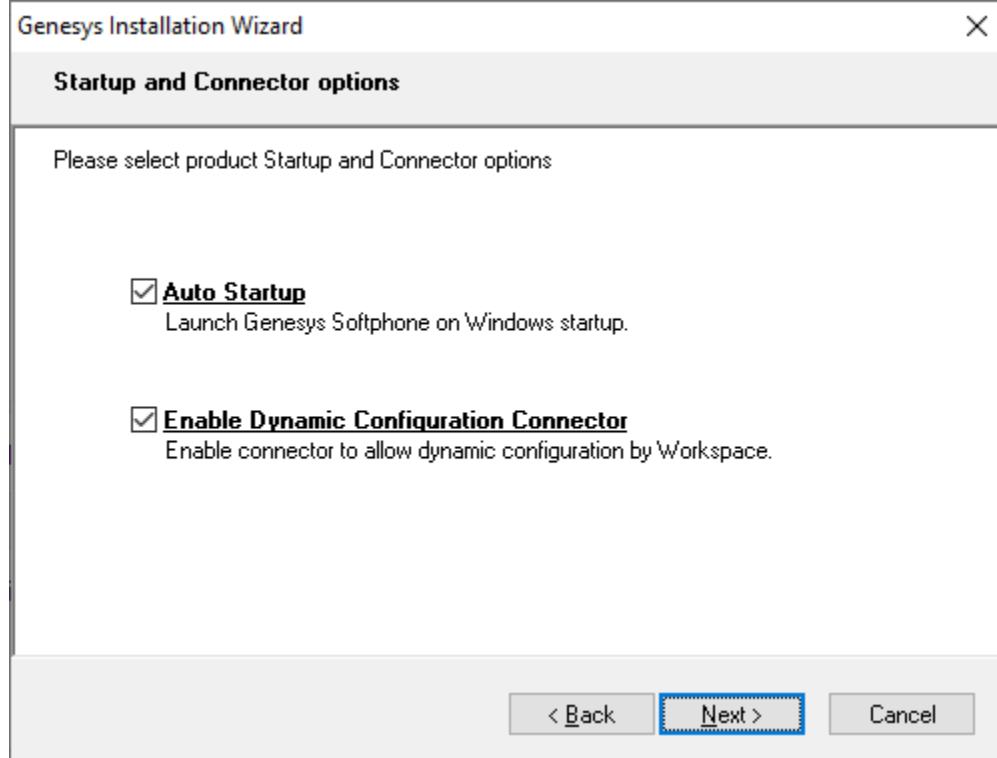
9. Click **Save**.
10. Go to **Desktop Options > Genesys Softphone** section.
11. Select the following check boxes:
 - **Usage of Genesys Softphone**
 - **Uri** with https://localhost:8000
12. Configure other options if needed.
13. Click **Save**.

Configure Genesys Softphone in WebRTC mode

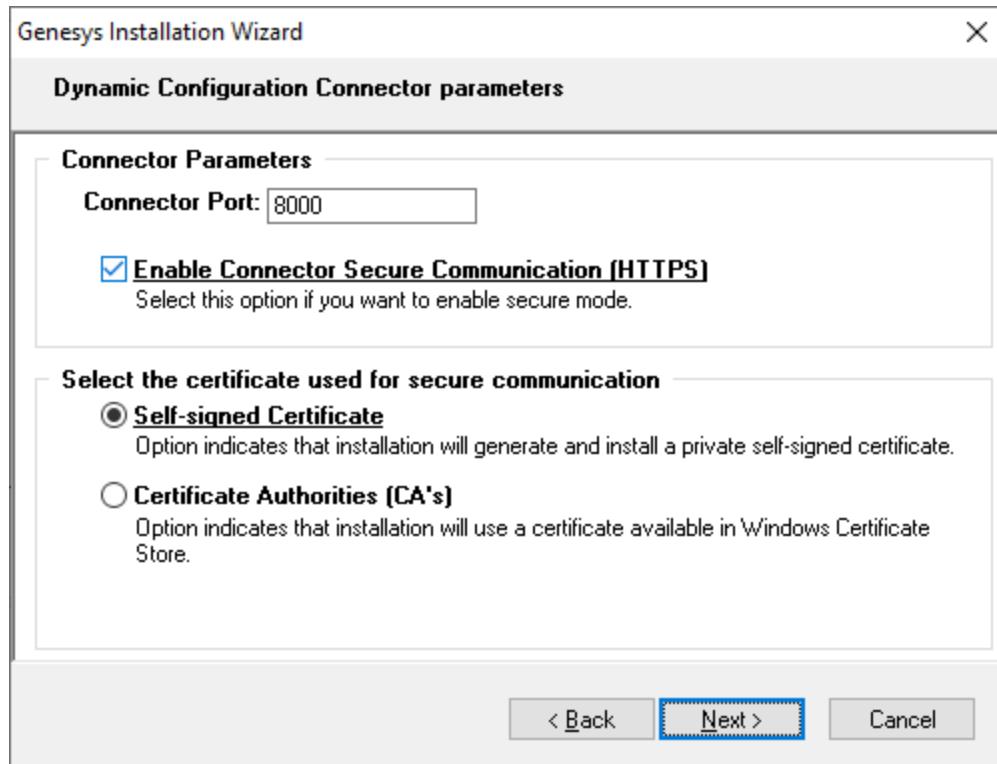
This section includes information on how to configure Genesys Softphone in WebRTC Connector mode.

Enabling Dynamic Configuration Connector in Connector Mode

1. Select **Auto Startup** and **Enable Dynamic Configuration Connector** checkboxes in the **Startup and Connector options** window.



2. Provide a proper connector port and select the **Enable Connector Secure Communication (HTTPS)** checkbox.



3. Select the **Self-signed Certificate** check box if you do not have appropriate certificate available in the Windows Certificate Store. Else, select the **Certificate Authorities (CA's)** check box. Proceed with the regular installation process.

Configure WebRTC agent with browser-based WWE via Agent Setup Application

Contents

- [1 Supported browser versions](#)
- [2 Configuration steps](#)

Agent does not need a Genesys Softphone as an endpoint for this configuration. But only needs a browser-based WWE, which uses WebRTC capabilities of Chrome, Firefox, and Chromium browsers.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

Supported browser versions

- Chrome - 70+
- Firefox - 62+
- Chromium - 85.0.564.41+

Configuration steps

To configure the WebRTC agent, perform the following steps.

1. Log in to the Agent Setup application via Genesys Portal for your Tenant. See Agent Setup guide on how to use Agent Setup application.
2. Select **Users** tab.
3. Click **Add User**.
4. Update all the mandatory fields:
 - First Name
 - Last Name
 - Username
 - Password
 - Password Confirm
 - Phone Number
5. Click **Save**.
6. Go to **Desktop Options > Voice**.

7. Select the **Can Use WebRTC** option.
8. Configure **Expression to capture groups in GWS url** and **WebRTC Service URN** options, if required.
9. Click **Save**.

The following table includes the options and its values for WebRTC configuration.

| Option Name | Option Value |
|---|---|
| Expression to capture groups in GWS url | It is a regular expression that allows the workspace to extract some part of its URL to capture the groups containing shared information among services, like the tenant or the region. |
| WebRTC Service URN | It is a WebRTC service URL that can be templated with the groups captured using the Expression to capture groups in GWS url option. |

Configure WebRTC Agents with Webphone

Contents

- 1 Configure WebRTC agent with Webphone using Agent Setup
 - 1.1 Provision Webphone
 - 1.2 Operate Genesys Webphone
 - 1.3 Operate WWE Agent Workspace
 - 1.4
 - 1.5 Control calls using Webphone and WWE

Webphone is a Browser-based standalone WebRTC softphone that can be used as the Agent's device separately from WWE wherever VDI (Citrix, VMware) is used.

Webphone is similar to Zero footprint that provides the ability to separate voice and signaling traffic.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

Configure WebRTC agent with Webphone using Agent Setup

To configure WebRTC agents with Webphone, perform the following configurations.

Provision Webphone

The following steps include the configuration for provisioning a Webphone,

1. Log in to **Agent Setup**.
2. Go to **Users > Add Users** to create an agent. To know more about user creation, see [Manage agents and other users](#).
3. Enter the mandatory fields to create the user.
4. Navigate to the configuration of the new agent.
5. Enter the agent's phone number and select the Softphone type as **Genesys Softphone / Genesys 420HT / AudioCodes 4xxHD / Polycom**.
6. Go to **User > Access Group**.
7. Configure the Access Group for the Agent. To know more about configuring access groups, see [Access Groups](#).

Important

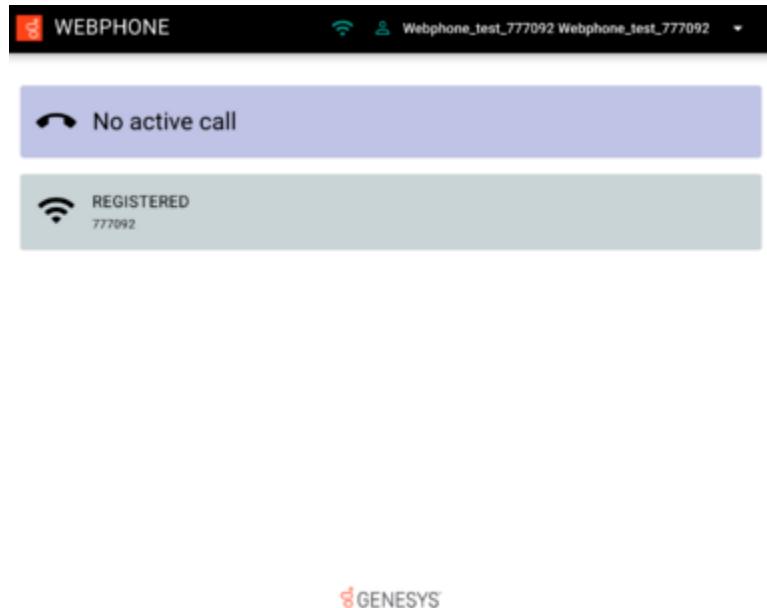
Make sure that the Access Group assigned to a new user has permissions to access the Folder where the new user is created. You can view the Folder in the **General Info** tab mentioned in the user creation process.

8. Go to **Desktop Options > Voice**.

9. Clear the **Can Use WebRTC** check box.
10. Click **Save**.

Operate Genesys Webphone

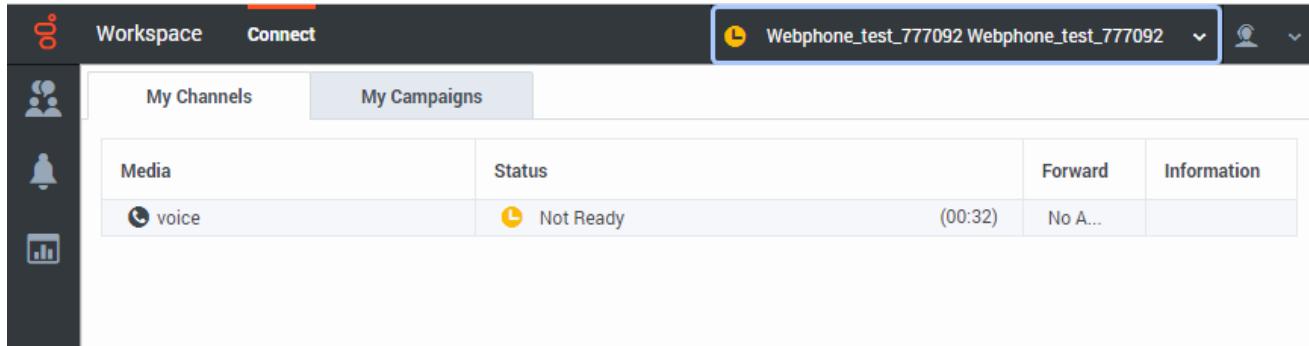
Log in to the Webphone using the credentials you have created. Log in to your Webphone as a WWE Agent now. You can view the following screen when the Webphone is successfully configured and connected to the Webphone service.



Operate WWE Agent Workspace

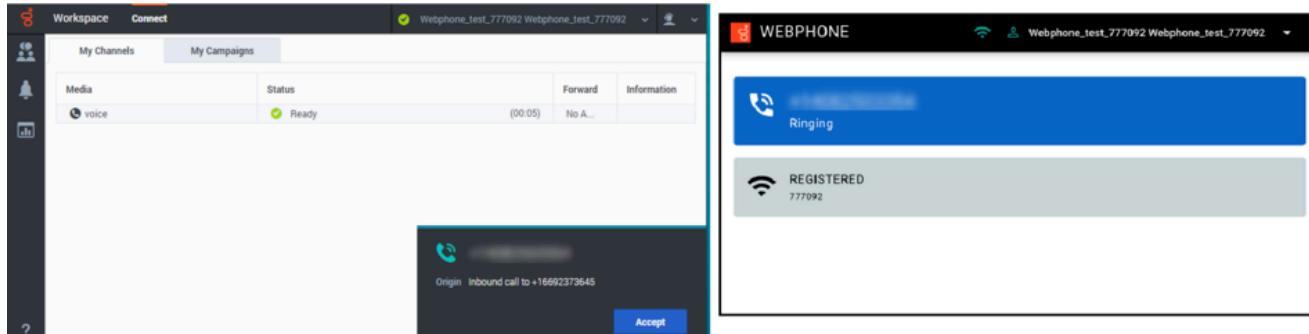
Log in to WWE with the user credentials that you created when configuring a WebRTC agent and you can view the following screen to initiate the Webphone call.

Configure WebRTC Agents with Webphone

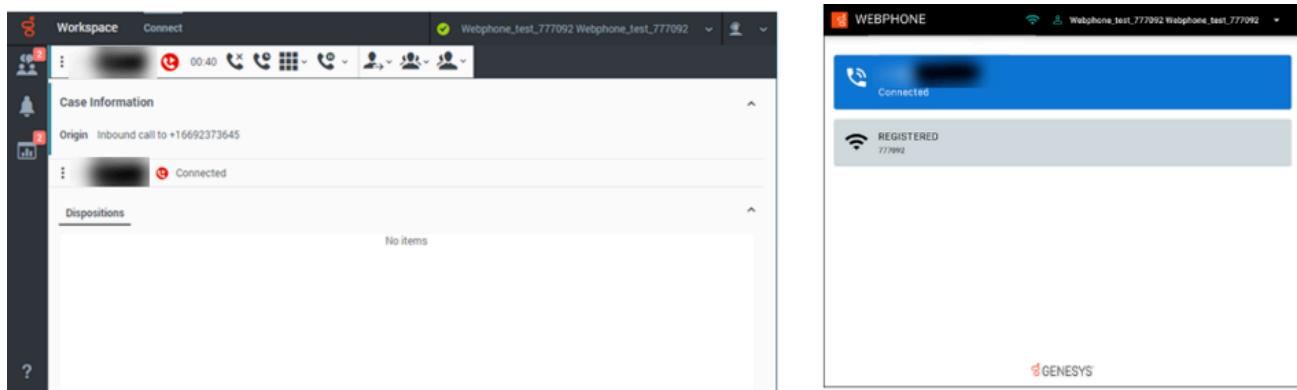


Control calls using Webphone and WWE

The agent can use the WWE desktop for all call control operations as the Webphone does not have any call control interface. The following image displays the inbound call ringing screen with WWE and Webphone.



The following image displays the screens once the call is established with WWE and Webphone.



Observability in WebRTC

Contents

- [1 Monitoring](#)
 - [1.1 Enable monitoring](#)
 - [1.2 Configure metrics](#)
- [2 Alerting](#)
 - [2.1 Configure alerts](#)
- [3 Logging](#)

Learn about the logs, metrics, and alerts you should monitor for WebRTC.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

Monitoring

Private edition services expose metrics that can be scraped by Prometheus, to support monitoring operations and alerting.

- As described on Monitoring overview and approach, you can use a tool like Grafana to create dashboards that query the Prometheus metrics to visualize operational status.
- As described on Customizing Alertmanager configuration, you can configure Alertmanager to send notifications to notification providers such as PagerDuty, to notify you when an alert is triggered because a metric has exceeded a defined threshold.

The services expose a number of Genesys-defined and third-party metrics. The metrics that are defined in third-party software used by private edition services are available for you to use as long as the third-party provider still supports them. For descriptions of available WebRTC metrics, see:

- [WebRTC Gateway Service metrics](#)

See also [System metrics](#).

Enable monitoring

The WebRTC service uses a PodMonitor custom resource definition (CRD). Monitoring is not enabled in the WebRTC service by default. To enable monitoring and expose WebRTC metrics and alerts, you must modify the Helm chart values. Set the following parameters in the **values.yaml** file to true:

- monitoring.enabled
- monitoring.prometheusMetrics
- monitoring.prometheusAlerts

For information about overriding Helm chart values before deployment, see [Overriding Helm chart values](#).

| Service | CRD or annotations? | Port | Endpoint/ Selector | Metrics update interval |
|------------------------|---------------------|-------|--------------------|-------------------------|
| WebRTC Gateway Service | PodMonitor | 10052 | /metrics | 30s |

Configure metrics

No further configuration is required in order to define or expose these metrics.

Alerting

Private edition services define a number of alerts based on Prometheus metrics thresholds.

Important

You can use general third-party functionality to create rules to trigger alerts based on metrics values you specify. Genesys does not provide support for custom alerts that you create in your environment.

For descriptions of available WebRTC alerts, see:

- WebRTC Gateway Service alerts

Configure alerts

Private edition services define a number of alerts by default (for WebRTC, see the pages linked to above). No further configuration is required.

The alerts are defined as **PrometheusRule** objects in a **prometheus-rule.yaml** file in the Helm charts. As described above, WebRTC does not support customizing the alerts or defining additional **PrometheusRule** objects to create alerts based on the service-provided metrics.

Logging

Refer to the Logging topic for information on configuring logging for WebRTC.

WebRTC Gateway Service metrics and alerts

Contents

- [1 Metrics](#)
- [2 Alerts](#)

Find the metrics WebRTC Gateway Service exposes and the alerts defined for WebRTC Gateway Service.

| Service | CRD or annotations? | Port | Endpoint/Selector | Metrics update interval |
|------------------------|---------------------|-------|-------------------|-------------------------|
| WebRTC Gateway Service | PodMonitor | 10052 | /metrics | 30s |

See details about:

- WebRTC Gateway Service metrics
- WebRTC Gateway Service alerts

Metrics

WebRTC exposes many Genesys-defined as well as system metrics. You can query Prometheus directly to see all the available metrics. The metrics documented on this page are likely to be particularly useful. Genesys does not commit to maintain other currently available WebRTC metrics not documented on this page.

| Metric and description | Metric details | Indicator of |
|---|---|--------------|
| wrtc_current_signins Specifies the number of current registered DNs | Unit: Type: Integer Label: Sample value: 2 | Monitoring |
| wrtc_current_in_calls Specifies the number of current incoming calls | Unit: Type: Integer Label: Sample value: 2 | Monitoring |
| wrtc_current_out_calls Specifies the number of current outgoing calls | Unit: Type: Integer Label: Sample value: 5 | Monitoring |
| wrtc_current_audio_calls Specifies the number of current audio calls | Unit: Type: Integer Label: Sample value: 5 | Monitoring |
| wrtc_current_video_calls | Unit: | Monitoring |

| Metric and description | Metric details | Indicator of |
|--|---|--------------|
| Specifies the number of current video calls | Type: Integer Label: Sample value: 2 | |
| wrtc_current_xcoding_calls Specifies the number of current xcoding calls | Unit: Type: Integer Label: Sample value: 2 | Monitoring |
| wrtc_peak_in_calls Specifies the maximum number of incoming calls | Unit: Type: Integer Label: Sample value: 50 | Monitoring |
| wrtc_peak_out_calls Specifies the maximum number of outgoing calls | Unit: Type: Integer Label: Sample value: 50 | Monitoring |
| wrtc_peak_audio_calls Specifies the maximum number of audio calls | Unit: Type: Integer Label: Sample value: 50 | Monitoring |
| wrtc_peak_video_calls Specifies the maximum number of video calls | Unit: Type: Integer Label: Sample value: 50 | Monitoring |
| wrtc_peak_xcoding_calls Specifies the maximum number of xcoding calls | Unit: Type: Integer Label: Sample value: 50 | Monitoring |
| wrtc_total_in_calls Specifies the total number of incoming calls | Unit: Type: Counter Label: Sample value: 100 | Monitoring |
| wrtc_total_out_calls Specifies the total number of outgoing calls | Unit: Type: Counter Label: Sample value: 100 | Monitoring |
| wrtc_total_audio_calls Specifies the total number of audio calls | Unit: Type: Counter Label: Sample value: 100 | Monitoring |
| wrtc_total_video_calls | Unit: | Monitoring |

| Metric and description | Metric details | Indicator of |
|--|--|--------------|
| Specifies the total number of video calls | Type: Counter Label: Sample value: 100 | |
| wrtc_total_xcoding_calls Specifies the total number of xcoding calls | Unit: Type: Counter Label: Sample value: 100 | Monitoring |
| wrtc_unauthorized_access Specifies number of unauthorized access attempts | Unit: Type: Counter Label: Sample value: 20 | Monitoring |
| wrtc_unknown_request Specifies the number of unknown requests received | Unit: Type: Counter Label: Sample value: 20 | Monitoring |
| wrtc_double_signin Specifies the number of registration requests that was received for registered DN | Unit: Type: Counter Label: Sample value: 20 | Monitoring |
| wrtc_rtp_losts Specifies the number of lost RTP packets | Unit: Type: Counter Label: Sample value: 20 | Monitoring |
| wrtc_rtp_errors Specifies the number of RTP receive errors | Unit: Type: Counter Label: Sample value: 2 | Monitoring |
| wrtc_rtp_gateway_jitter {over="100"} Audio quality monitoring metrics | Unit: Type: Counter Label: {over="100"} Sample value: | Monitoring |
| wrtc_rtp_gateway_jitter Audio quality monitoring metrics | Unit: Type: Counter Label: {over="300"} Sample value: | Monitoring |
| wrtc_rtp_gateway_jitter Audio quality monitoring metrics | Unit: Type: Counter Label: {over="500"} Sample value: | Monitoring |
| wrtc_rtp_client_jitter | Unit: | Monitoring |

| Metric and description | Metric details | Indicator of |
|--|---|--------------|
| Audio quality monitoring metrics | Type: Counter Label: {over="100"} Sample value: | |
| wrtc_rtp_client_jitter Audio quality monitoring metrics | Unit: Type: Counter Label: {over="300"} Sample value: | Monitoring |
| wrtc_rtp_client_jitter Audio quality monitoring metrics | Unit: Type: Counter Label: {over="500"} Sample value: | Monitoring |
| wrtc_system_error Specifies the number of failed ICE transactions | Unit: Type: Integer Label: {type="turn_errors"} Sample value: | Error |
| wrtc_system_error Specifies the number of registration transactions which were timed out | Unit: Type: Integer Label: {type="sips", sip=""} Sample value: 2 | Error |
| wrtc_system_error Specifies if WebRTC is able to connect to Elasticsearch server or not | Unit: Type: Integer Label: {type="es"} Sample value: 1 or 0 | Error |
| wrtc_system_error Specifies the number of error responses received from Elasticsearch server | Unit: Type: Counter Label: {type="es_errors"} Sample value: 2 | Error |
| wrtc_system_error Specifies if WebRTC is able to connect to GAuth service or not | Unit: Type: Integer Label: {type="auth"} Sample value: 1 or 0 | Error |
| wrtc_system_error Specifies the number of error responses received from GAuth server | Unit: Type: Counter Label: {type="gauth_errors"} Sample value: 2 | Error |
| wrtc_system_error Specifies if WebRTC is able to connect to GWS Configuration service or not | Unit: Type: Integer Label: {type="cfg"} Sample value: 1 or 0 | Error |
| wrtc_system_error | Unit: | Error |

| Metric and description | Metric details | Indicator of |
|--|--|--------------|
| Specifies the number of error responses received from GWS Configuration server | Type: Counter Label: {type="cfg_errors"} Sample value: 2 | |
| wrtc_system_error Specifies if WebRTC is able to connect to GWS Environments service or not | Unit: Type: Integer Label: {type="env"} Sample value: 1 or 0 | Error |
| wrtc_system_error Specifies the number of error responses received from GWS Environments service | Unit: Type: Counter Label: {type="env_errors"} Sample value: 2 | Error |
| wrtc_max_clients_per_instance Specifies the maximum number of clients per instance | Unit: Type: Constant Label: Sample value: | Performance |
| wrtc_max_clients_per_node Specifies the maximum number of clients per node | Unit: Type: Constant Label: Sample value: | Performance |
| wrtc_calls_by_domain Specifies the number of calls by domain | Unit: Type: Counter Label: Sample value: | Performance |
| wrtc_registrations_by_domain Specifies the number of client registrations per domain | Unit: Type: Counter Label: Sample value: | Performance |
| wrtc_failed_registrations_by_domain Specifies the number of failed client registrations per domain | Unit: Type: Counter Label: Sample value: | Performance |
| wrtc_client_errors Specifies the number of double sign-ins being rejected | Unit: Type: Counter Label: (type = double_sign_in_reject) Sample value: | Error |
| wrtc_client_errors Specifies the number of clients being dropped on sign-out | Unit: Type: Counter Label: (type = dropped_on_signout) Sample value: | Error |
| wrtc_client_errors | Unit: | Error |

| Metric and description | Metric details | Indicator of |
|---|--|--------------|
| Specifies the number of client timeout errors | Type: Counter Label: (type = timeout) Sample value: | |

Alerts

The following alerts are defined for WebRTC Gateway Service.

| Alert | Severity | Description | Based on | Threshold |
|------------------------|----------|---|----------------------|------------|
| webrtc-gateway-signins | warning | Specifies the number of sign-ins | wrtc_current_signins | 15mins |
| webrtc-gateway-gauth | warning | Specifies that the Gateway Pod has lost connection to Auth service | wrtc_system_error | Need input |
| webrtc-gateway-gws | warning | Specifies that the Gateway Pod has lost connection to the Environment Service | wrtc_system_error | Need input |
| webrtc-gateway-es | warning | Specifies that the Gateway Pod has lost connection to ElasticSearch | wrtc_system_error | Need input |

Logging

Contents

- 1 [Gateway](#)
- 2 [CoTurn](#)

Learn how to store logs for WebRTC.

Related documentation:

-
-
-

RSS:

- [For private edition](#)

Gateway

Log files are mandatory for troubleshooting process. WebRTC creates logs in both logfiles and Elasticsearch. Logs are created either using HostPath or PersistentVolumeClaim parameters.

Use the following formula to calculate the approximate log storage volume:

$(\text{Number of Agents}) * (\text{Average call load per hour}) * (0.2 \text{ constant}) * (\text{Number of active hours})$

As all operations are of the same order, it can be scaled up or down linearly. For example, if 500 GB is required for 1000 agents with 10 calls per hour for 24 hours, then for 2000 agents with the same load and time we need around 1TB (both are with a slight buffer for idling time).

CoTurn

Coturn creates logs in logfiles or stdout. Logs are created either using HostPath or PersistentVolumeClaim parameters.

A full day (24H) of logging for a 1000 connected agents will give us the required capacity of around 100 GB of storage, considering that we have a buffer for standardization. For easy calculation, the capacity can be scaled linearly from 100 GB for more number of agents. Use the following formula for the exact calculation:

$3.6\text{MB/hr} * \text{number of hours} * \text{number of agents}$